Norphonic Heavy Duty VoIP Telephones

for Industrial and Emergency Areas

Manual

Contents

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1. General

The Norphonic Heavy Duty VoIP Telephone is a robust, weather resistant telephone built to withstand extreme temperatures, humid and dirty environments.

The Norphonic Heavy Duty VoIP telephone is approved for use in emergency and industrial areas, with an overall weather resistance up to IP65. The unit is manufactured in a durable aluminum casing, resistant to corrosion, rust and condensation. The hookswitch has no moving parts, making the telephone ideal for use in environments exposed to high levels of dust or grime and comes complete with a 80 dB ringer as standard.

For secure operation, the unit features built-in selftests of the internal switch and telephone components. The test results can be transmitted to a central surveillance system. Optional extenal call-in test facility is also provided. The internal state (port state, hook off, selftest) can be read by external control systems over Modbus UDP, Modbus TCP and SNMP.

Norphonic can be ordered with an optional built-in ethernet switch featuring two single mode fibreoptic sockets (100Mbit type LC connectors) which supports redundant networking via RSTP. The switch supports QoS (Quality of Service), differentiating between high and low priority traffic in convergent networks. This makes the telephone an ideal communication unit for voice and data traffic.

The Norphonic telephone is otherwise available with or without keypad and with or without with a flashing light beacon for use in noisy areas where a ring may not be heard. The model without keypad is used to initiate a PABX hotline/ Autodial on handset lift, or wait for reply.

1.1. About This Manual

Please review this document before mounting and installing your new Norphonic Heavy Duty VoIP Telephone. This Manual contains set-up information, accessories overview for mounting of the telephone as well as guidelines for cable connections and fittings (LAN, Power and Fiber cable connections)

This Manual is valid for the following Norphonic Heavy Duty VoIP Telephone models:

- **Norphonic TRA-K1-F1**
  With Keypad, with single mode fiber ports

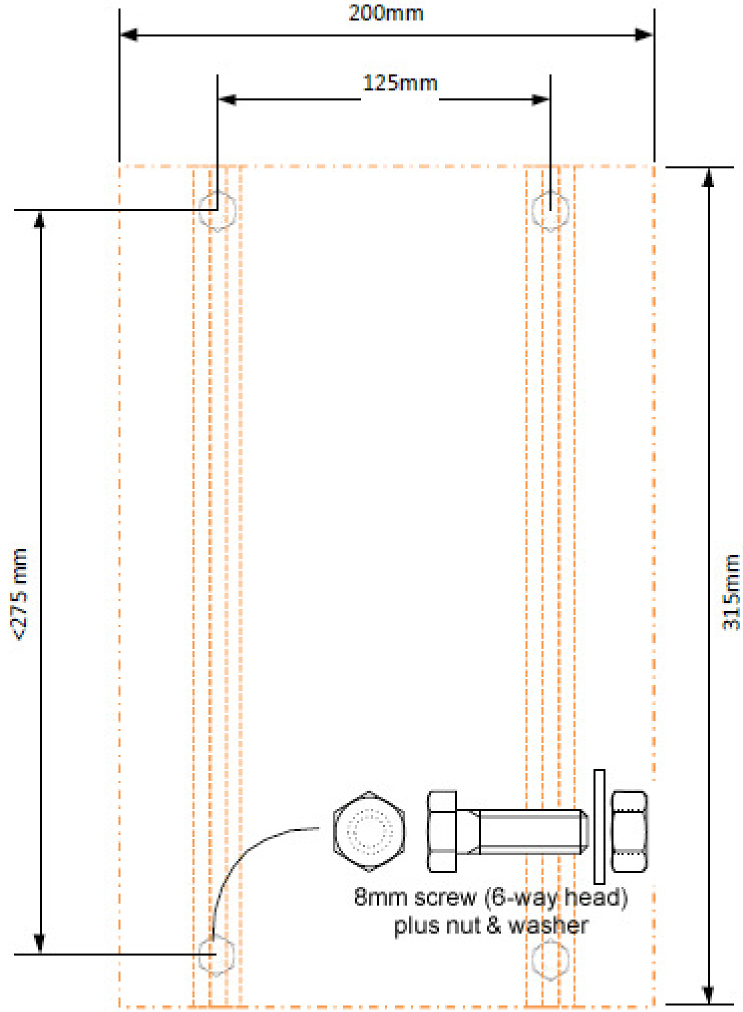
- **Norphonic TRA-K1-F0**
  With Keypad, without single mode fiber ports

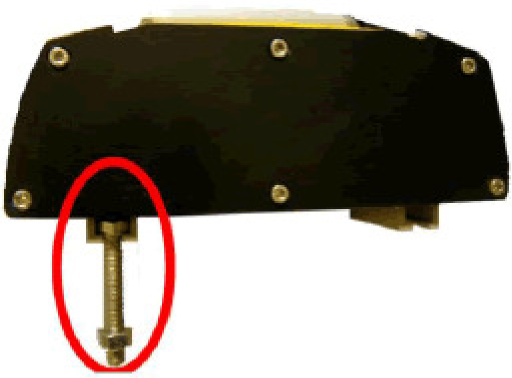
- **Norphonic TRA-K0-F1**
  Without Keypad, with single mode fiber ports

- **Norphonic TRA-K0-F0**
  Without Keypad, without single mode fiber ports

2. Telephone Mounting and Cable Connections

2.1 Mounting - With Access to the Back (ie door etc)

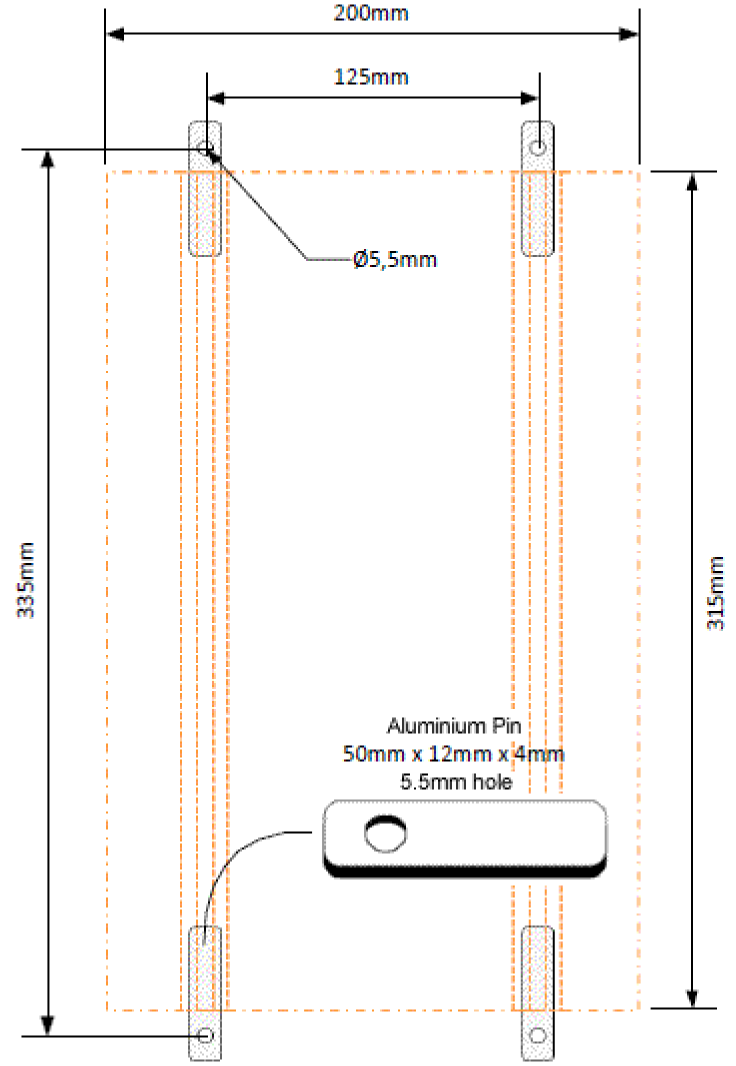


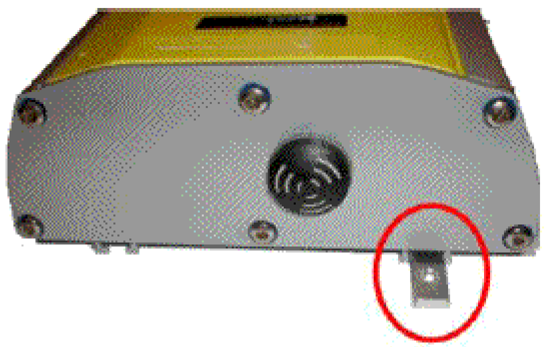


At the back of the telephone, there are two profiles as shown on the picture.

- An 8mm screw with a 6-way head will fit into these profiles.
- It is 12.5cm between the profiles at the back of the telephone
- Recommended mounting in a cabinet door or plate (with an easy access to the back) is:
    - 4 units of a 8mm screw with 6-way head + nut and metal washer
    - The lenght of the screw is 16mm.
    - The metal washer and nut is mounted at the back of the closet door or plate.
    - *(Note: «mounting accessories – screws, nuts & washers» are available through Norphonic.)*

2.2. Wall Mounting of Norphonic



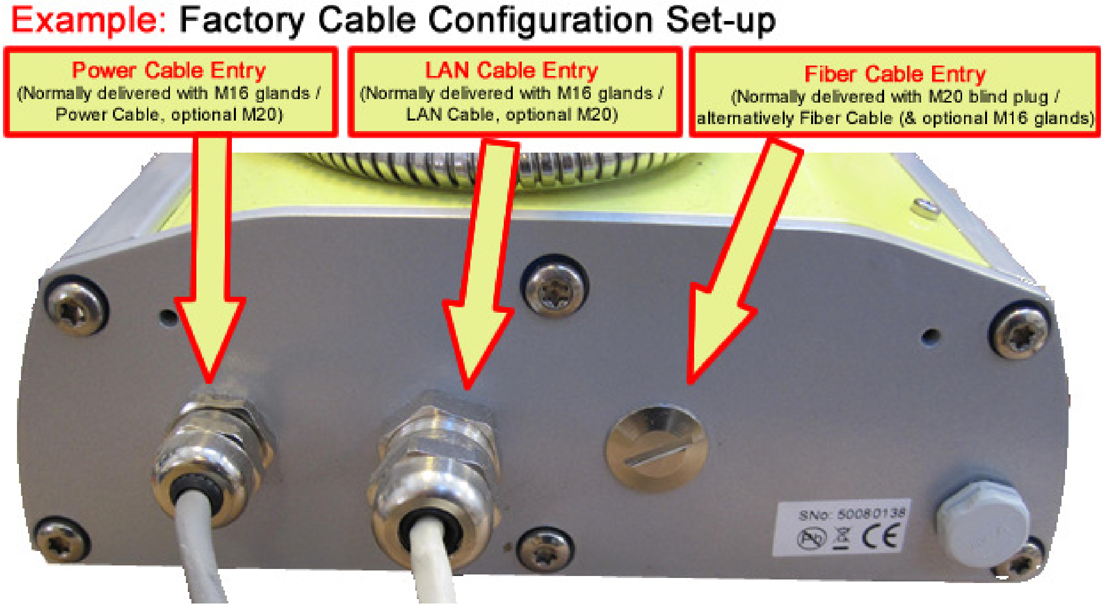


When the Norphonic unit is mounted on a wall:

- Two (2) aluminum pins (50x12x4mm) are used for each telephone. These fits into the profile at the back of the telephone (shown on the picture).
- The aluminum pins are mounted diagonally (for example one at the upper right hand side and one at the bottom left hand side)
- If the telephone is mounted at a wall with bad mounting grip, it is recommended to use 4 aluminum pins for each telephone.

Note: *«mounting accessories - pins» (Aluminum pins) can be ordered separately through Norphonic.*

2.3. Cable Accessories, Dimensions and Specifications



2.3.1. Power Cable Specifications

The Norphonic Heavy Duty VoIP Telephone comes with a 2 metre long Power cable as standard (24V DC). This Power cable comes complete with a two pins Power connector (IP67). A separate datasheet on the power cable is available upon request. Note that the Power cable is fitted to the telephone via M16-sized Glands.

Optional Configurations Available (extras):

- 5 metre or 10 metre lenght Power cables are also available, or any special order.
- The Norphonic telephone could also be customised with the larger M20-sized Glands.

2.3.2. LAN Cable Spesifications

The Norphonic Heavy Duty VoIP Telephone comes with a 2 metre long a high quality S-FTP type LAN cable as standard. A separate LAN cable datasheet is available upon request. Note that the LAN cable is fitted to the telephone via a M16-sized Glands.

Optional Configurations Available (extras):

- 5 metre or 10 metre lenght LAN cables are also available, or any special order.
- The Norphonic telephone could also be customised with the larger M20-sized Glands.

2.3.3. Fiber Cable Spesifications

The standard Norphonic Heavy Duty VoIP Telephone comes with a M20-sized blind plug fitted in the Fiber Cable entrance. If you have ordered a telephone with two single mode fiber ports / 100Mbit type LC connectors, we can also deliver the fiber cable & connections by special request:

Optional Configurations Available (extras):

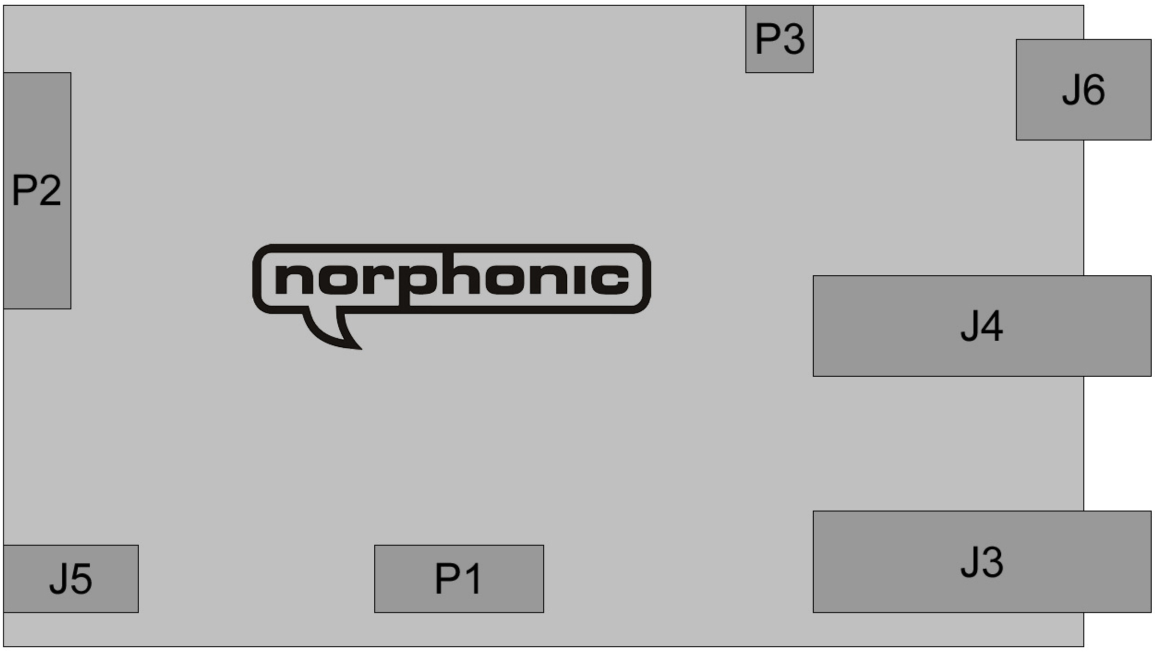
- 2 metre, 5 metre or 10 metre lenght Fiber cables are available, or any special order.
- The Norphonic telephone could be customised with the smaller M16-sized Glands.

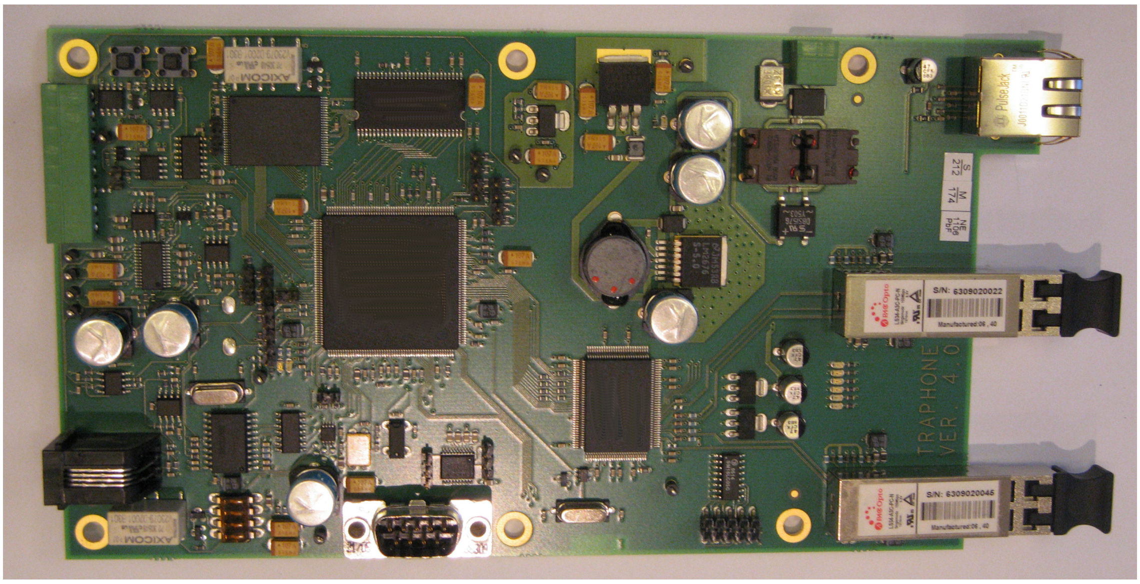
3. Technical Features

Ethernet switch	1 x 10/100BaseTX Twisted pair, RJ45 socket. 2 x 100BaseFX Multimode or Singlemode fibre, LC socket. Supports QoS/ToS (Quality of Service/Type of Service).
Self test	Link state detection on all switch ports. Local self test of telephone, including microphone and speaker.
Telephony/VoIP	Transport: Session Initiation Protocol (RFC 3261) Session Description Protocol (RFC 2327) Real-time Transport Protocol (RFC 1889) Voice Codec: G.711 (A-law and Mu-law)
Data	Integrated Modbus/UDP and Modbus/TCP slave for access to internal status (link status, hook on/off, telephone components state).
Body	IP 65.Extruded aluminium IP 65.Extruded aluminium main body.
Handset	IP 66.Robust handset with stainless steel cord.
Operating Temperature	÷25 - +55 degrees Celsius
CE	The telephone meet the requirements and protection guidelines of the following EC Directives and comply with the harmonized European standards (EN) issued in the Official Journal of the European Communities with regard to radio and telecommunications terminal equipment: - 89/336/EEC "Electromagnetic Compatibility" (EMC Directive) - 1999/5/EC "Radio and telecommunication terminal equipment" (RTTE Directive)
Power	24Vdc, normally 0.35 A max. 0.5 A

4. Telephone Installation

4.1 Connectors





P1	Programming Internal port for programming/debugging of cpu
P2	IO Input for hook off detection RS485 data bus Line out for external speaker relays for external signalling 1. Hook off (Reed relay) 2. Hook off (Reed relay) 3. NC 4. NC 5. NC 6. NC 7. RL1 (Buzzer relay) 8. RL2 (Buzzer relay)
P3	Power supply 24V DC
J3	Ethernet fibre port (optional) 100Base-FX LC Multimode or singlemode
J4	Ethernet fibre port (optional) 100Base-FX LC Multimode or singlemode
J5	Telephone handset 1. MIC 2. SPEAKER 3. SPEAKER 4. MIC
J6	Ethernet copper port 10/100 Base TX twisted pair, RJ45 socket

4.2 Configuration

When the telephone boot it will request IP address, Subnet mask, Gateway address and Boot server host name (option 66) using DHCP, then it will request configuration data from the specified TFTP server. If it fails to receive IP/configuration settings it will start up with the previous configuration (stored in flash).

The configuration is stored in a text file '<ip>.cfg' or '<mac>.cfg' where <ip> is the assigned IP address, and <mac> is the MAC address of the phone. If both files exist, telephone will pick up configuration from <ip>.cfg.

Example of a configuration file:

```
# SIP display name and user name for this phone.
sip_display_name=MyName
sip_user_name=1001

# Proxy server location and user name.
sip_proxy_server=10.0.0.15:5060
proxy_user_name=1001
proxy_domain_name=

# If dial_enabled=0, off-hook event will call the following hotline receiver
hotline_receiver=999@10.0.0.15

# Network setup for the phone, should normally never be changed.
default_if_name=br0
transport_tcp=false
sip_port=5060

# Audio setup for the phone, should normally never be changed.
audio_device=/dev/dsp
audio_frag_size=128
audio_frag_count=4
audio_aec_settings=0x5490 0x2a82 0xff0a

# Handset volume setting, in percent
audio_volume=80

# Selftest and proxy reregistration intervals
selftest_interval=1800
reregister_interval=3600

cli_auth_user_name=
cli_auth_domain_name=
cli_auth_passwd=

srv_auth=false
srv_auth_user_name=
srv_auth_realm=
srv_auth_passwd=

# Enable keypad by setting dial_enabled=1, and set numbering schema if required.
dial_enabled=0
#dial_numbering_schema=:8;
#dial_domain=10.0.0.15
```

4.3 Customizable settings

These settings can be changed by the end user.

Parameter	Description
sip_display_name	The display name of the unit, can be arbitrary and include whitespaces.
sip_user_name	The user name of the unit, must be unique and include no spaces.
sip_proxy_server	Proxy server name or IP address.
proxy_user_name	User name for registration with proxy server.
proxy_domain_name	Domain of the proxy.
hotline_receiver	SIP URI for connection in emergency telephone mode.
selftest_interval	Time between audio self-tests in seconds.
reregister_interval	Time between registration requests to proxy. Different vendors may have varying demands to minimum interval.
dial_enabled	0 if the unit operates in emergency mode, 1 otherwise
dial_numbering_schema	Prefix numbering schema for dialing; see description below.
dial_domain	The domain part to form SIP URI together with dialed number.
dial_timeout	The delay after which the typed in number is dialed
test_caller_id	Caller ID for audio loopback / TUF-test
audio_volume	Handset audio volume level, on the scale 1 to 100. Defaults to 80.

4.4 System settings

These settings must not be changed by end user.

Parameter	Description
audio_device	Device used for audio output. Shall not be modified.
audio_frag_size	Audio playback fragment size. Shall not be modified.
audio_frag_count	Number of the fragments. Shall not be modified.
audio_aec_settings	Initialisation settings for audio echo canceller.

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4.5 Dialing rules

There are two ways of entering valid number sequence, via dialing-in the number and waiting for timeout or by using pre-defined set of dialing rules.

The optional field dial_timeout specifies the delay in seconds after last keypress to commit a call. Default is 4 seconds. To disable this feature, 0 should be specified.

The field dial_numbering_schema contains a list of semicolon-separated dialing rules. Each entry is in form of <prefix>:<remainder-length>, where prefix is the sequence to dial in order to match the entry, and remainder-length is a number specifying how many digits are expected after the prefix. For example, a rule for making an international call to Norway could be 0047:8.

The prefixing defaults to no prefix and 4 digits, however it is useful to include it explicitly as a reminder. The order of dialing rules is not important. If prefixes share common beginning, the most specific one dialed will be used to select a rule, e.g. with both '0:14', '00:16' and '0047:8' defined, the latter will be used if '0047' was dialed in.

The line describing above would be:

dial_numbering_schema=:4;0:14;00:16;0047:8

Once a session is established, inband DTMF postfixes can be used.

Telephones can also operate in emergency mode, with SIP recipient defined by hotline_receiver field. In this case, no dialing is required.

4.6 TUF test

In addition to built-in self-test, telephones provide facility for remote self test. To enable it, specify test_caller_id configuration file. When the telephone is called from source matching the specified caller ID, it will pick up the call and establish audio loop for testing. Then, a test sound can be played and heard back by test system. When remote party hangs up, the unit returns to normal operation.

If the option is omitted or empty, the test facility is disabled.

4.7 SNMP Support

Telephones support SNMP management by running an SNMP agent on board. The units share same configuration file, snmpd.conf. It should be placed in TFTP root directory and is fetched automatically upon start-up.

The monitoring software should be able to receive SNMP notifications decribed in TRAPHONE-MIB.txt. Currently provided notifications are:

- **unitFailure** A notification generated when telephone software specific failure occurs. The variable passed contains specific message describing failure.
- **hookStatus** A notification generated when hook goes off or on. The variable passed contains the new status of telephone hook.

- **proxyRegistrationStatus** A notification generated when unit registration alters. The variable passed contains the specific status message.
- **selfTestStatus** A notification generated when unit DTMF self-test result alters. The variable passed contains the specific status message.

Additionaly, an **updateStatus** variable is defined. The variable is a trampoline to run an update by writing in an IP address of update server, and provides status check for update progress and completion by reading from it.

A number of non telephone specific monitoring definitions are supported, namely those defined in MIB2 and UCD-SNMP frameworks.

The configuration file should be written per NET-SNMP project snmpd.conf file format. In particular, trapsink to the SNMP monitoring host should be specified.

4.8 Remote Update

Telephone firmware can be remotely updated to newer versions. There are two ways to perform an update:

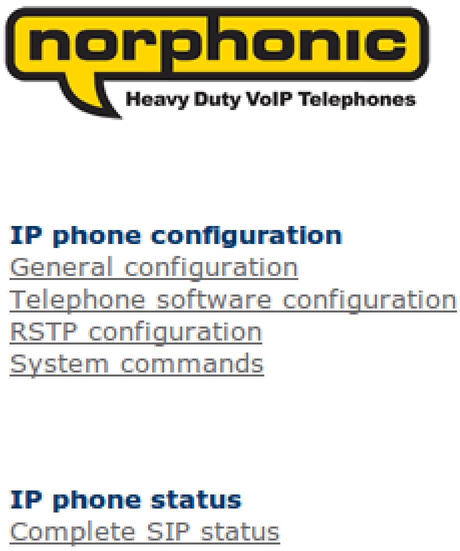
- Initiate by writing IP address of TFTP server containing updates to updateStatus SNMP variable. The update will begin immediately and its progress will be reflected in the variable.
- Scheduled update. Every 24 hours each telephone unit checks the TFTP server listed in DHCP option 66 for new firmware version. Once found, an update commits. The updates are scheduled at different time for each unit to avoid bottlenecks in networks with massive deployment.

In both cases, TFTP server should contain the following files:

- **zImage** Linux kernel image
- **md5.zImage** A checksum for Linux kernel image
- **tpupdate.tgz** An archive with updated userland sofware and configuration files
- **md5.tpupdate.tgz** A checksum for the archive
- **fw_version** A file containing the version number of new firmware

5. Web Interface

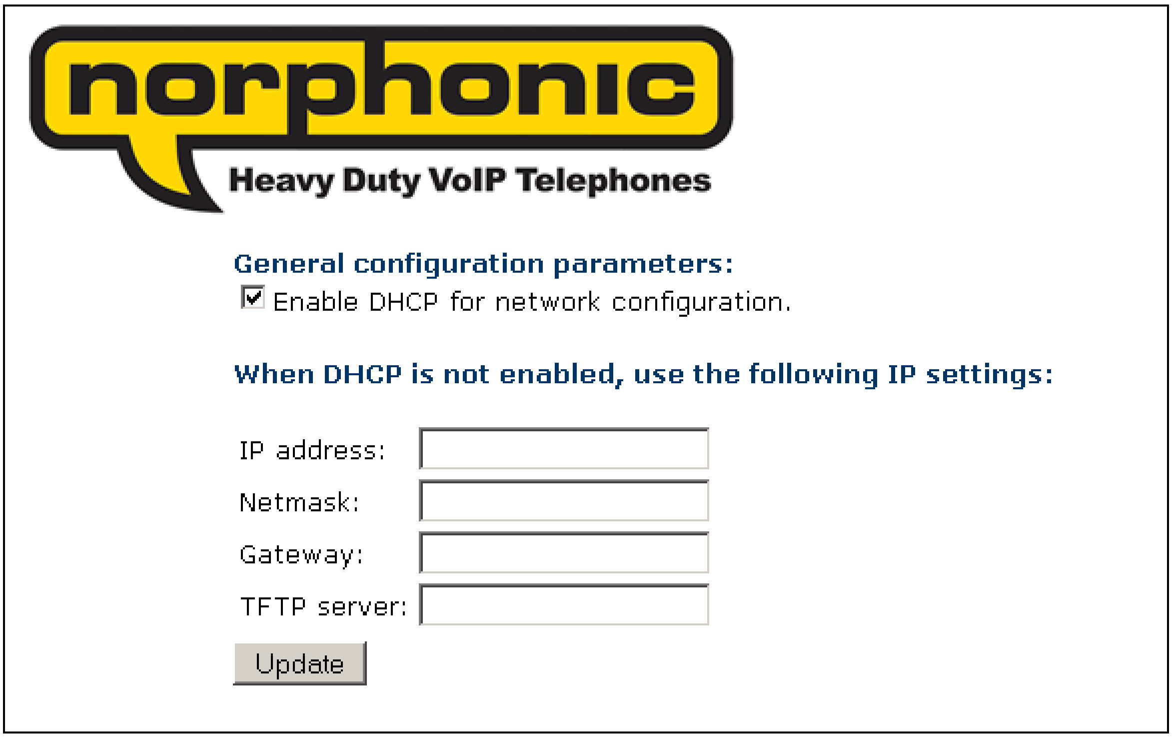
Telephones also provide web interface for configuring and monitoring unit's operation. They have simple password-protection with username **phoneadmin** and password **blank**.



Main menu of the web interface

5.1 General configuration page

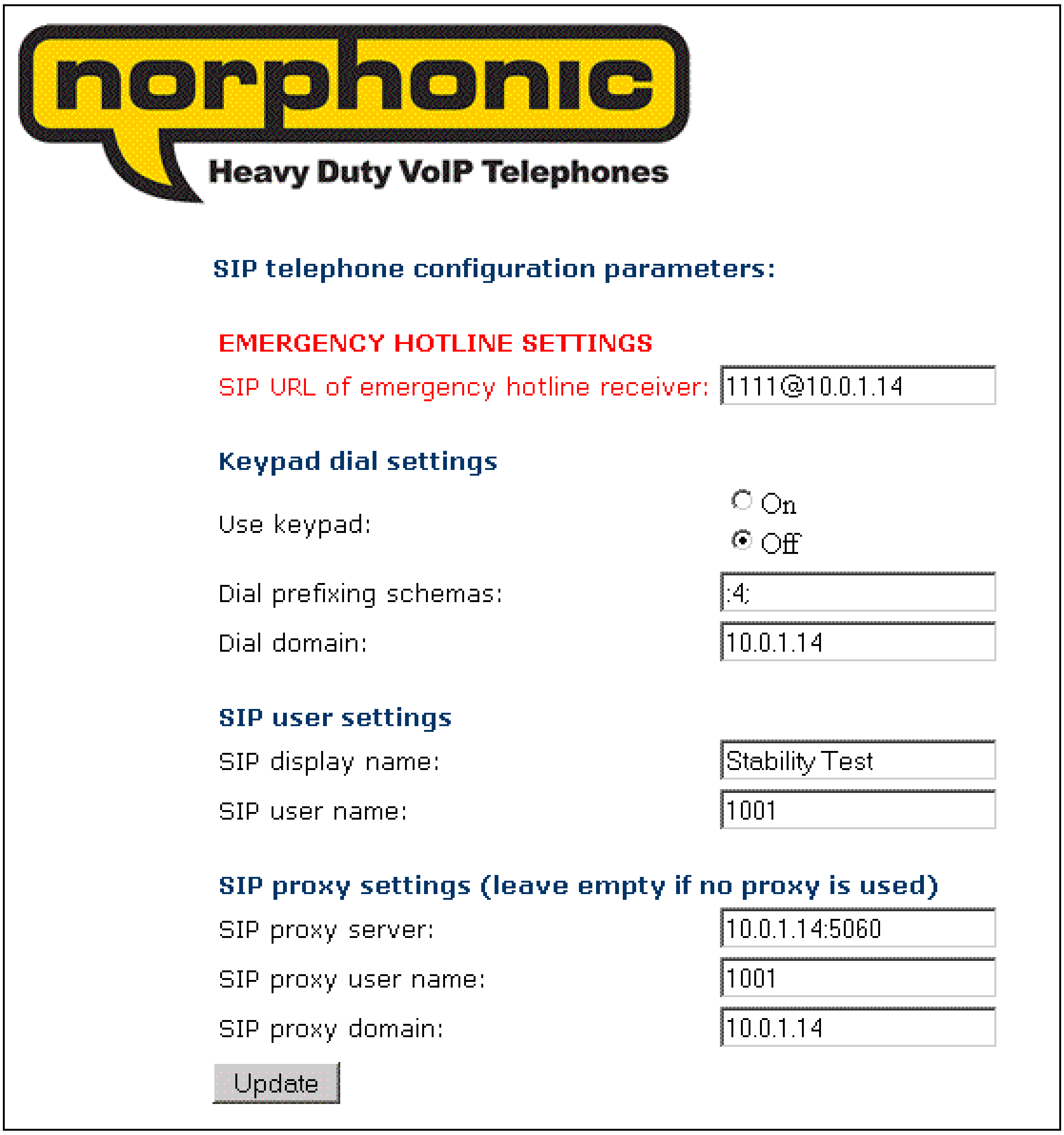
General configuration presents user with unit's general network settings. If no DHCP protocol selected, static values of IP address, network mask, default gateway and TFTP server have to be specified.



General configuration menu

5.2 Telephone software configuration

Telephone software configuration gives the end user custom options as follows:

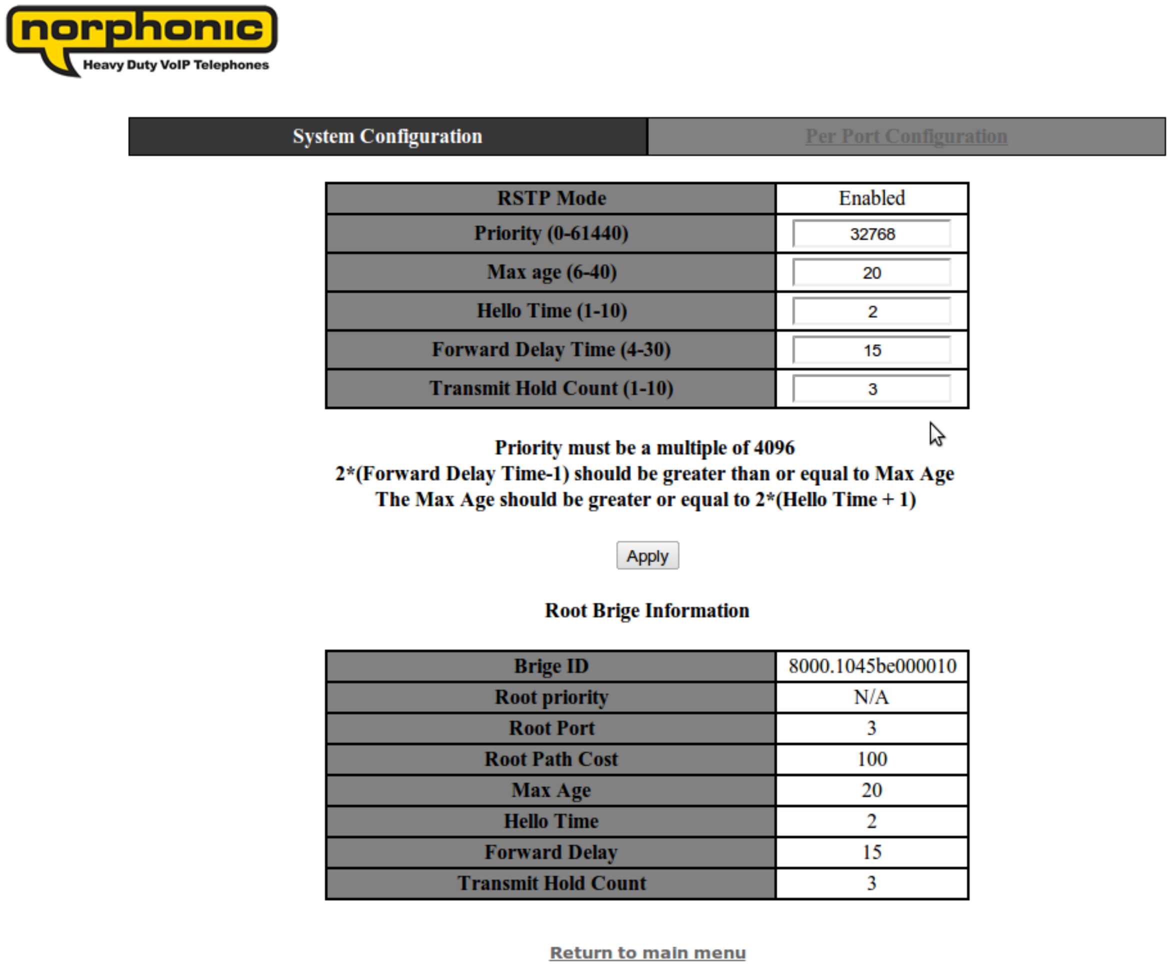


Telephony settings menu

5.3 RSTP Configuration

Telephones with RSTP support enabled provide a set of options to configure spanning tree settings. It is recommended that RSTP settings of the phones are matched with settings of other RSTP switching equipment present in the network segment.

The system configuration page allows to adjust RSTP bridge priority and protocol timings. It highlights the information about bridge ID, priority, root port and active timings.



norphonic Heavy Duty VoIP Telephones

System Configuration | Per Port Configuration

RSTP Mode	Enabled
Priority (0-61440)	32768
Max age (6-40)	20
Hello Time (1-10)	2
Forward Delay Time (4-30)	15
Transmit Hold Count (1-10)	3

Priority must be a multiple of 4096
2*(Forward Delay Time-1) should be greater than or equal to Max Age
The Max Age should be greater or equal to 2*(Hello Time + 1)

Apply

Root Brige Information

Brige ID	8000.1045be000010
Root priority	N/A
Root Port	3
Root Path Cost	100
Max Age	20
Hello Time	2
Forward Delay	15
Transmit Hold Count	3

Return to main menu

RSTP system configuration menu

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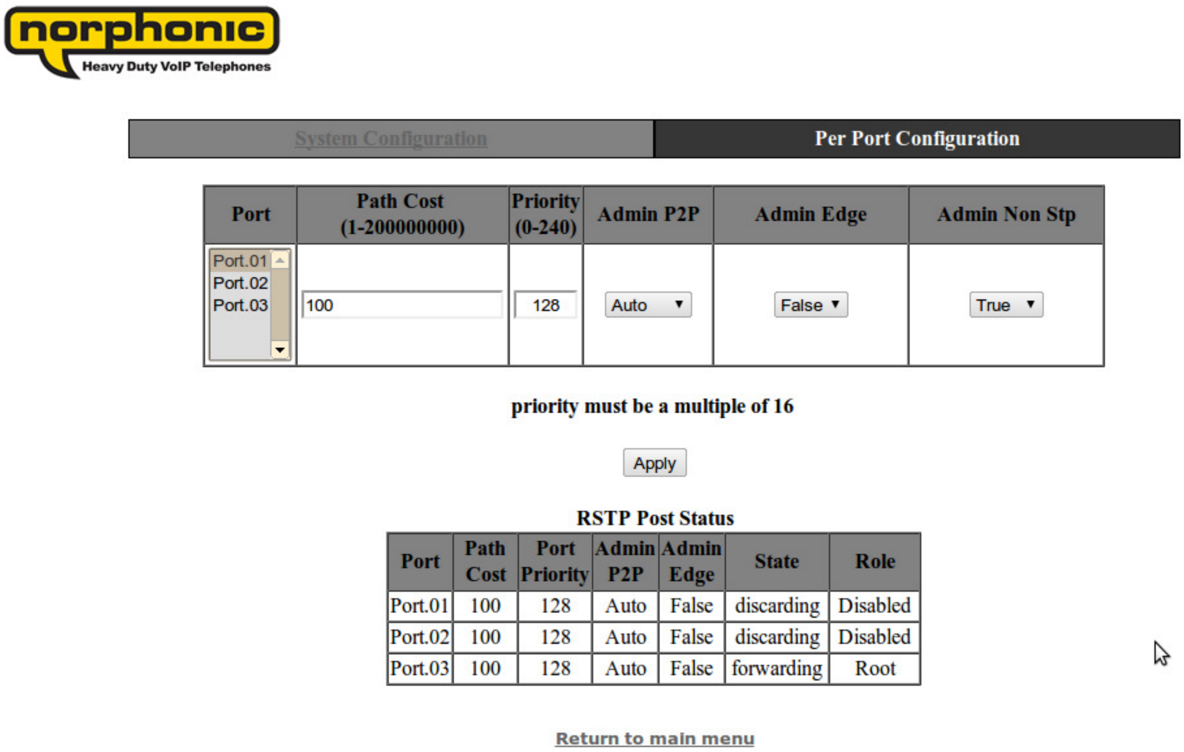


Port	Path Cost	Port Priority	Admin P2P	Admin Edge	State	Role
Port.01	100	128	Auto	False	discarding	Disabled
Port.02	100	128	Auto	False	discarding	Disabled
Port.03	100	128	Auto	False	forwarding	Root

RSTP per port configuration menu

Per port configuration menu allows to ajdust settings relevant to each physical network port of the phone. Ports number 1 and 2 are fiberobtic ports; port 3 is copper CAT5.

User can adjust path cost, port priority, edge, P2P and STP admin settings. The active settings of the ports are reflected in table at the bottom.

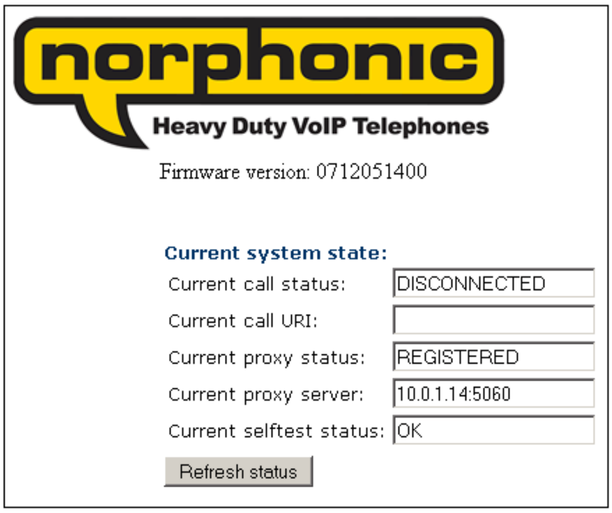
5.4 System commands

System commands provide two options: restarting onboard telephony software or rebooting the whole system.

System commands menu

5.5 Complete SIP status

commands Complete status page allows to monitor current unit status (self-test, proxy, call status) and also reflects the installed firmware version.



Current system status menu

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6. Data communication (Modbus)

Modbus UDP and Modbus TCP
Supports Read multiple registers (FC 3) and Write multiple registers (FC 16)

		1	1	1	1	1	1	0	0	0	0	0	0	0	0	0	0
Addr	Name	5	4	3	2	1	0	9	8	7	6	5	4	3	2	1	0
512	State	-	-	-	-	-	-	b	a	-	-	-	-	-	p2	p1	p0
513	Runcnt																

State, internal state of the phone
a- 1=hook off/0=hook on
b- 1=seltest OK/0=selftest FAILED
p0- Port 0, 1=link OK (Fiberoptic port)
p1- Port 1, 1=link OK (Fiberoptic port)
p2- Port 2, 1=link OK (RJ-45)

Runcnt, internal runcounter that increase each time the telephone is polled

7. Technical data

7.1 Audio characteristics

Audio codecs:
- G.711 (A-law and Mu-law), G.729, G.729B

Audio quality:
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Line Echo Cancellation (LEC)
- Audio Echo Cancellation (AEC)
- Automatic Gain Control (AGC)
- Packet Loss Concealment (PLC)
- Jitter buffer (Fixed, sequential, adaptive)

Audio tones:
- Progress tone
- Dial tone
- Busy tone
- Ring tone

7.2 Telephony services

SIP:
- Place or cancel outgoing calls
- Accept or reject incoming calls
- Register with SIP registrar

7.3 Configuration:

Network:

- DHCP or static IP
- Static IP address
- Subnet mask
- Gateway IP address
- MAC address
- DNS IP address

SIP:

- SIP server and port number
- Outbound SIP proxy server and port number
- SIP display name and username
- SIP authentication userid and password
- Call forward SIP URLs
- SIP Registration interval
- Local SIP listen port

Media:

- Local RTP port
- DTMF method (inband)
- VAD enabled or disabled
- Preferred packet time
- Configurable list of audio codecs

Misc:

- APS server name
- Hot dial-up number

7.4 Networking

Protocols:

- SIP - Session Initiation Protocol (RFC 3261)
- SDP - Session Description Protocol (RFC 2327)
- RTP - Real-time Transport Protocol (RFC 3550)
- DNS client with A and SRV records
- DHCP client (RFC 1541)
- UDP - User Datagram Protocol (RFC 0768)
- IPv4 - Internet Protocol version 4 (RFC 0791)
- IEEE 802.3 Ethernet
- RSTP – Rapid Spanning Three Protocol

Interfaces:

- Switched Ethernet 10/100 Mbit/s
- 100BaseFX Multi/Single mode LC

7.5 Interoperability

The SIP UA has been tested with SIP proxies and registrars such as:

SIP Proxies;

- Alcatel A5020 Proxy and B2BUA
- Asterisk PBX
- Avaya proxy
- Cisco
- Hotsip Sapphire
- Partysip
- Radvision Prolab
- SIP Express Router (SER)
- Snom 4S
- Ubiquity proxy
- Vovida Vocal
- Yokogawa TTB3010 IPv4/IPv6 translator
- Brekeke OnDO SIP Server
- Brekeke OnDO PBX

SIP User Agents:

- Audiocodes ATA
- Cisco 186 ATA
- Cisco 7960 SIP phone
- Cisco-SIPGateway/IOS-12.x
- Grandstream Handytone and Budgetone
- KPhone
- Microsoft Messenger
- Mitel 5055
- Pingtel v2.2.0
- Radcom
- Radvision SIP toolkit v3.1
- SjPhone
- Snom 100 and Snom 200
- Tecom IP2006
- X-lite
- Yamaha RT56v
- Siemens SCS-Client Professional 1.1.0
- Vegastream 50 BRI

Other:

- Navtelcom SIP UA+Proxy simulator
- Netscreen 500 ALG
- Protos test suite

8. Confidentiality notice

9. Contact Norphonic

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