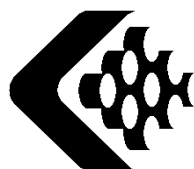
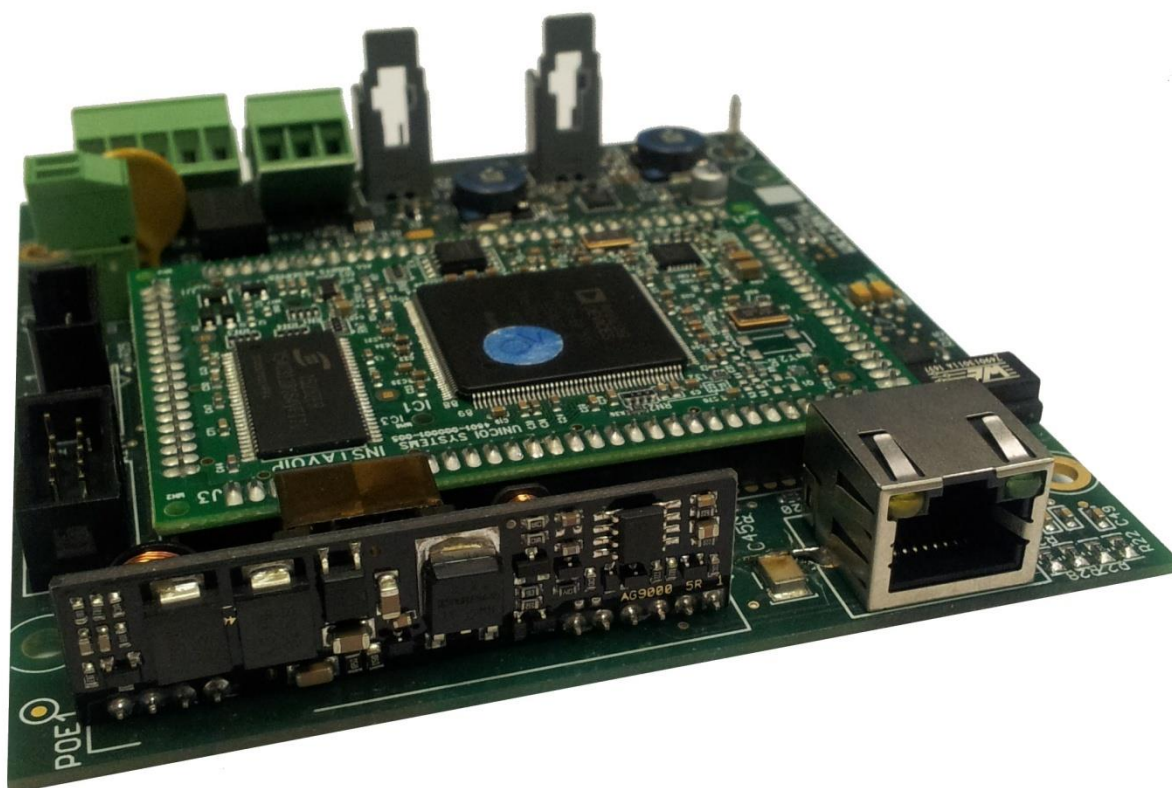


# INSTALLATION AND USE INSTRUCTIONS

## IP / POE INTERCOM NETWORK

### INTERCOM CARD - WK120NTH



**LE LAS Group**  
COMMUNICATION UNDER HIGH PROTECTION

## **- SUMMARY -**

<b><u>1. PRESENTATION OF YOUR WORKSTATION'S WEB SERVER</u></b> .....	<b>6</b>
1.1 ENTER THE SERVER.....	6
1.2 NAVIGATE THE SERVER.....	7
<b><u>2. DESCRIPTION OF YOUR WORKSTATION'S WEB SERVER</u></b> .....	<b>8</b>
2.1 ACCOUNTS .....	8
2.2 AUDIO .....	10
2.3 NETWORK .....	13
2.4 SYSTEM .....	15
2.5 MANAGEMENT .....	16
2.6 ADVANCED .....	17
2.7 DIRECTORY .....	21
<b><u>3. GRANDSTREAM PBX SERVER PROGRAMMING</u></b> .....	<b>22</b>
<b><u>4. BASIC USE OF A CHEF STATION</u></b> .....	<b>23</b>
4.1 CLASSIC COMMUNICATIONS .....	23
4.2 GENERAL CALL AND GROUP CALL .....	24
4.3 CONFERENCE .....	25
4.4 CARRYOVER.....	25
4.5 DOUBLE CALL.....	25
4.6 MINI-CONFERENCE .....	26
4.7 TRANSFER.....	26
4.8 MISSED CALLS.....	26
<b><u>5. ADVANCED FEATURES OF A CHEF STATION</u></b> .....	<b>27</b>
5.1 PUSH TO TALK.....	27
5.2 DIRECTORY .....	28
5.3 SENDING DTMF CODES .....	28
5.4 STRIKE OPTION .....	28
5.5 TIME LIMITATIONS .....	29
5.6 SPEAKER VOLUME.....	29
5.7 CALL PRIORITIES .....	29
5.8 MANAGING 2 SIP ACCOUNTS .....	30
5.9 STATION RESET .....	31
<b><u>6. BASIC USE OF A SECONDARY STATION</u></b> .....	<b>31</b>
6.1 CLASSIC COMMUNICATION .....	31
6.2 GENERAL CALL AND GROUP CALL .....	32
<b><u>7. ADVANCED FEATURES OF A SECONDARY STATION</u></b> .....	<b>33</b>

## **Equipment for industrial sites**

### **IP / PoE INTERCOM CARD**

#### **WK120NTH**

#### **ATTENTION**

BEFORE PROCEEDING WITH COMMISSIONING, READ CAREFULLY  
THE MANUAL PROVIDED TO ENSURE THAT THE CONFIGURATION  
FACTORY CORRESPONDS TO THE DESIRED USE.

#### **REMARKS**

THE WARRANTY ONLY APPLIES UNDER THE INSTALLATION CONDITIONS  
AND OPERATION STRICTLY DEFINED IN THIS MANUAL.  
NO GUARANTEE CAN BE IMPLEMENTED  
IF DETERIORATION RESULTS FROM A FOREIGN CAUSE  
TO THE DEVICE OR NON-COMPLIANCE WITH THE INSTRUCTIONS FOR USE.  
FOR THE SAKE OF IMPROVEMENT, THE INFORMATION CONTAINED IN  
THIS DOCUMENT AND THE CHARACTERISTICS OF THE MATERIALS MAY  
SUBJECT TO CHANGE WITHOUT NOTICE.

#### **EUROPEAN STANDARDS**

UNITS MARKED "CE" COMPLY WITH THE DIRECTIVE  
EMC (2014/30/EU),  
TO THE LOW VOLTAGE DIRECTIVE (2014/35/EU)  
FORMULATED BY THE EUROPEAN COMMUNITY.

## **- Technical characteristics -**

*12V or POE power supply.*

*Max consumption 1A at 12V, 12W*

*Audio output 10W Class D amplifier under 8 Ohm speaker.*

*Hands-free or handset audio output full duplex or half duplex push to talk.*

*Management of local echo and ambient noise suppression.*

*Codecs: G.711 ulaw, G.711 alaw, G.722 HD, G.726 (16kbps), G.726 (24kbps), G.726 fixed payload, G.726 (40kbps), DVI4 Narrowband, DVI4 HD, Linear PCM, Linear PCM HD, Linear PCM (little endian),*

*Linear PCM HD (little endian), ILBC-30, ILBC-20, SPEEX Narrowband, SPEEX Wideband, SILK Narrowband, SILK Wideband.*

*SIP, RTP, STUN, NTP.*

*Caller ID and Caller Name.*

*General call and group calls (requires LELAS SIP server).*

*Conference (requires LELAS SIP server).*

*Peer to Peer mode allowing point-to-point calls without a server (call by IP addressing).*

*Direct call by programmable memory key for secondary extensions (10 keys)*

*Communication visualization by indicator light (flashing: incoming and outgoing call, steady: communication in progress).*

*Management of a strike by DTMF code activation.*

*Report/transfer.*

*Setting on the station on the station's embedded web server: number of rings before answering, maximum online time, loudspeaker volume, ringtone and microphone, addition of personalized ringtones (ringtone and ring back), lock option, postponement/transfer number, IP and Netmask address, cascade dialling, direct call numbers, SIP settings, codecs, button press time (line connection, call cancellation), addition of three broadcast messages, etc.*

*2-line, 16-character LCD screen for head station, scrolling information.*

*Static or DHCP or PPPoE mode*

### **Possible option with LELAS SIP server**

*The SIP server system uses the UCM61xx from GrandStream with a LELAS configuration*

*Simultaneous calls depending on the server (from 30 to 60).*

*Support up to 500 SIP terminal registrations.*

*Supports up to 32 conference participants, conference invitation. 99 conference channels.*

*Supports queue mode by static agents with different strategies (simultaneous, Linear, least recent, fewer calls, random, Circular with memorization). Editable music on hold broadcast.*

*Call interception.*

*Voicemail, messaging consultation with confidential code.*

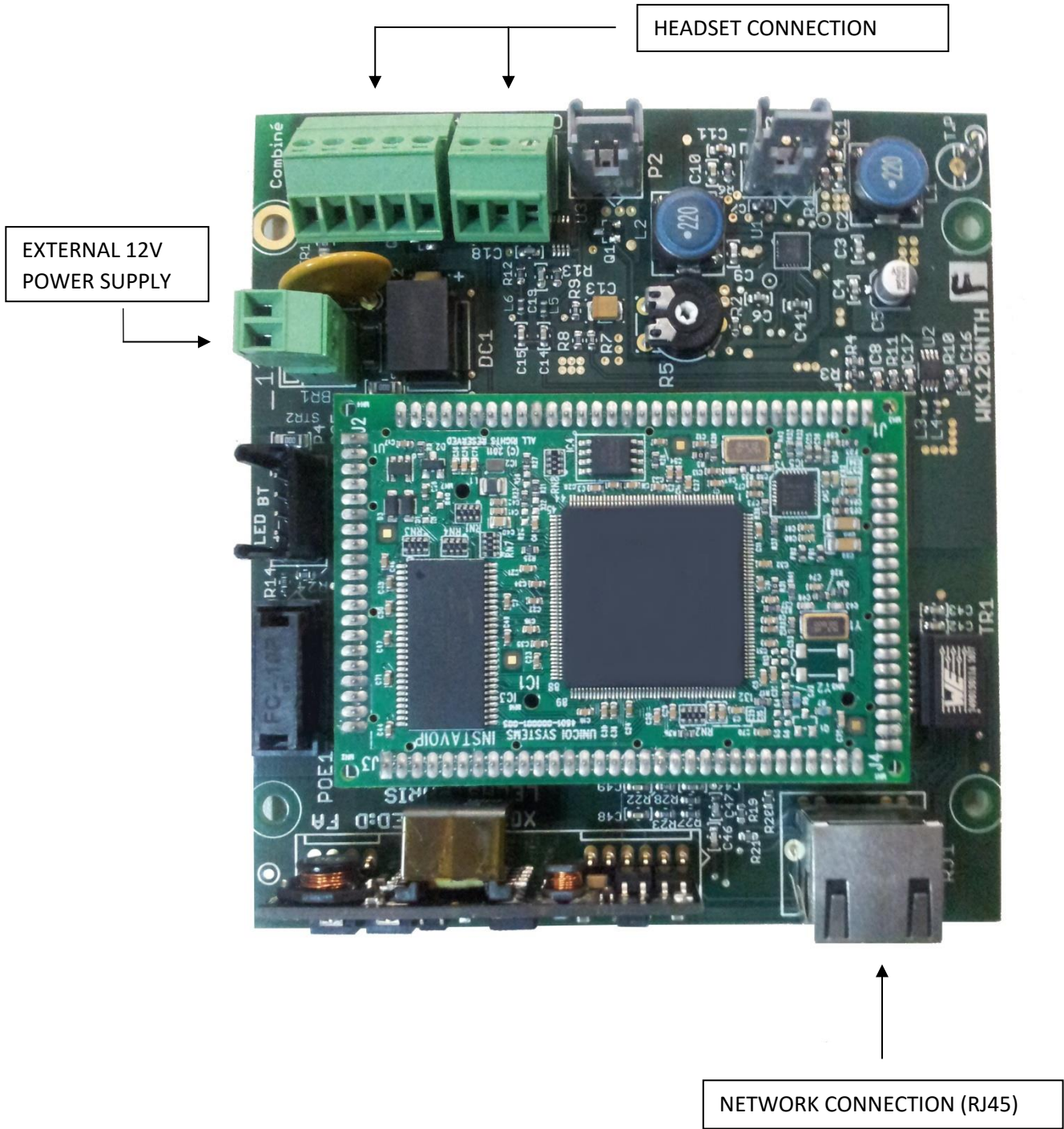
*Interactive voice response.*

*DND call forwarding (Do not disturb, call forwarding if busy, if no answer, unconditional..).*

*Recording of communications (WAV format) saved on SD card.*

*SIP server language: English, French, Chinese, Spanish, German, Russian, Italian*

# - WK120NTH CONNECTION -



# - INSTALLATION INSTRUCTIONS -

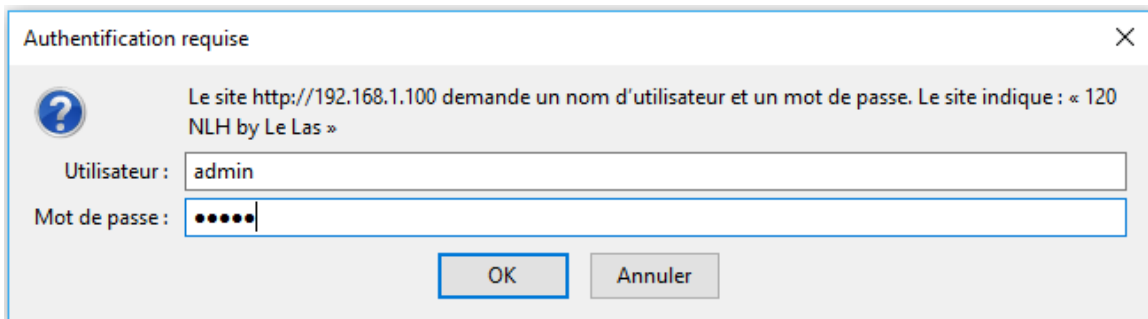
## 1. Presentation of your workstation's web server

### 1.1 Enter the server

Each workstation equipped with the WK120NTH card has its own web server. Accessible from its IP address, the server allows you to configure the entire extension: from fundamental parameters such as the creation of a SIP account, to the settings of each telephone function such as automatic answering of the extension, etc.

To access it, you need to know the IP address of the station. By default, stations use DHCP to obtain an address, however, it is strongly recommended for the stability of your local network to define a static address afterwards.

To find the IP address of each of your workstations, we advise you to use a network scanner, such as "IPScan". Once you know this address, enter it into your browser's address bar and press enter.



When you arrive on the position page, you will be asked for a username and password, by default "admin" / "admin". Enter them and press "OK".

You have now entered your workstation's web server and will be able to start configuring it.

## 1.2 Navigate the server

First, you will always find these four elements present on your web interface:

- (1) The navigation bar: this will be used to navigate between each workstation configuration page. Organized by theme, you will find, for example, the setting of the station volumes in the “Audio” tab.
- (2) The authentication controller: It allows you to automatically log out after a 10-minute countdown. “Renew” allows you to reset this countdown, and “Logout” allows you to log out manually.
- (3) The status bar: it allows you to display the status of your account (Register, Unregister, etc.), your IP address, your MAC address, the date and time. It also allows notifications to appear when one or more parameters have been modified and must be saved, or when parameters have been saved and the system needs to restart.

- (4) Page content: you will always find here the name of the page you are on and the theme of this page.



## 2. Description of your workstation's web server

### 2.1 Accounts

#### a. Page layout

This page is used to configure your VoIP account.

**Accounts**  
Add an account to connect to a PBX.

**(Unconfigured account)**

Account Actions:  
Disable  
Register  
Unregister

General | Topology | QoS | Advanced

Account Name  
Display Name  
Username/Number  
Domain  
Register with domain  
Password

Second account

**Status**  
(Unconfigured account)  
User: (Unconfigured account)  
Status: Account not configured

**System**  
IP: 192.168.1.188 (Static)  
MAC Address: 18-39-19-40-43-FE  
System time: 2011-10-02 13:44:06  
Uptime: 1d 2h 57m 26s

Time until auto-logout: 09:33

The SIP account configuration section has the following elements:

- (1) Account name: Your account name is displayed here.
- (2) Account actions: These three buttons allow you to control the status of the account.  
You can enable, disable, register or unregister it from your PBX server.
- (3) Account configuration bar: A SIP account has a large number of settings. We have organized them into several tabs, but for a basic configuration of your account, only the “General” page needs to be duly completed.
- (4) Content of the selected tab: Display of all the configurable fields of the tab.
- (5) Check box allowing you to create a second account on the station. Once created, you can deactivate it by unchecking this same box.



## b. Basic configuration of a SIP account

For the majority of users, configuring a SIP account is limited to the “General” tab; as for the other tabs, only modify the values if you really know what you are doing.

The image shows a configuration interface for a SIP account. At the top, there are four tabs: "General" (selected), "Topology", "QoS", and "Advanced". Below the tabs, there are several input fields and a checkbox:

- Account Name:** A text input field with a question mark icon to its right.
- Display Name:** A text input field with a question mark icon to its right.
- Username/Number:** A text input field with a question mark icon to its right.
- Domain:** A text input field with a question mark icon to its right.
- Register with domain:** A checkbox that is currently unchecked, with a question mark icon to its right.
- Password:** A text input field.

First, you can fill in the “Account Name” field with any name you want.

This name will only be used and visible on the server itself, so it doesn't matter much. Next, there are two different ways to fill in the fields, depending on whether you use a PBX server on your network or not:

- If you have a PBX server, please complete the following fields:
  - **Username/Number:** This field corresponds to the extension to which you have assigned this extension in your PBX server. It is therefore a number ranging from 100 to 999.
  - **Domain:** Here you have to enter the address of your PBX server. This can be the DNS name of the server (example: “sip.company.com”), or the numeric address (example: “192.168.1.23”).
  - **Register with domain:** You must absolutely check this box for your extension to register on your PBX server.
  - **Password:** You must enter here the password that you have associated with the extension chosen for this extension in your PBX server so that it can register correctly.
- If you do not have a PBX server, simply choose a name for your extension and enter it in the fields: Account Name, Display Name and Username/Number. Additionally, check that the “Register with domain” box is unchecked.

## 2.2 Audio

On this page you will be able to adjust the different volumes of the set, modify your ringtone and your call return tone, but also choose the codecs to use.

Accounts Audio Network System Management Advanced Directory Renew Logout

Time until auto-logout: 02:48

### Audio

Choose preferred codecs.

#### Volume Control

**Speaker**  
Full duplex  
Push to talk

**Microphone**  
Full duplex  
Push to talk

**Ringtone**  
Ringback

**GC**

Ring tone:  Default  
 Play ringtone.wav  
Select audio file Upload ring tone

Ringback tone:  Default  
 Play ringback.wav  
Select audio file Upload ringback tone

Scalable ringtone:

LS/Mic TEST

#### Codec Selection

Choose Preferred Codecs


Available	Preferred
G.711 uLaw	G.726 fixed payload
G.711 aLaw	DVI4 Narrowband
G.726 (16kbps)	G.711 uLaw
G.726 (24kbps)	G.711 aLaw
G.726 fixed payload	
G.726 (40kbps)	
DVI4 Narrowband	
Linear PCM	
Linear PCM (little endian)	
G.729	
ILBC-30	
ILBC-20	
SPEEX Narrowband	
SILK Narrowband	

Enable >> << Disable Move Up Move Down

### Status


**1000**  
User: 1000@192.168.1.36  
Status: Registered

**System**  
IP: 192.168.1.120 (Static)  
MAC Address: 18-39-19-41-C1-6C  
System time: 2011-10-01 06:54:05  
Uptime: 7m 25s

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### a. Volume adjustment

To modify a volume, simply move its slider to the right to increase it, or to the left to decrease it:

-  : Corresponds to the speaker volume. It will define the gain amplification of the received audio signal.


The first bar (**Full duplex**) corresponds to the desired volume when the station is communicating in “Full Duplex”.

The second bar (**Push to talk**) corresponds to the desired volume when the station is in “Push to talk” communication.

-  : Corresponds to the microphone's output volume. It will define the gain amplification of the signal transmitted on the network.

The first bar (**Full duplex**) corresponds to the desired transmission gain when the station is communicating in “Full Duplex”.


The second bar (**Push to talk**) corresponds to the desired transmission gain when the station is in “Push to talk” communication.

-  : Corresponds to the ringtone and ringback volumes.

The first bar (**Ringtone**) defines the volume with which the ringtone will sound.

The second bar (**Ringback**) sets the volume at which the ringback tone will sound. The ringing ringback tone is the one you hear when you call a remote extension and it is ringing.

- **GC** : Corresponds to the volume at which your general call will be heard.

-  : Corresponds to the listening volume of your telephone handset.

### b. Choice of ringtone and ringback tone

You can integrate your own ringtone and ringback sound.

**Attention:** downloaded files must be in .wav format, and not exceed 100KB.

Ring tone  Default  
 Play ringtone.wav

Ringback tone  Default  
 Play ringback.wav

To upload your sound, click on the associated “Select audio file” button, then choose your sound and click on “Upload ring tone” or “Upload ringback tone” depending on the tone you are modifying.

Finally, choose “Play ringback.wav” or “Play ringtone.wav” from the radio button selection, to play these sounds as a ringtone or as a ringback tone. If you leave “Default”, the default associated tone will sound.

### c. Evolutionary ringtone

Scalable ringtone :

By checking this box, you activate the progressive ringtone. This means that when your extension starts to ring, the volume of your ringtone will start at minimum, then increase every 3 seconds until reaching the volume defined on this same page (see 2.2.a).

### d. Codecs

A codec is a device for implementing the encoding or decoding of voice, with a view to transmitting it. Some codecs also include a compression or data encryption function.

**Codec Selection**  
 Choose Preferred Codecs

Available	Preferred
G.711 uLaw	DVI4 HD
G.711 aLaw	DVI4 Narrowband
G.726 (16kbps)	G.711 uLaw
G.726 (24kbps)	
G.726 fixed payload	
G.726 (40kbps)	
G.722 HD	
DVI4 Narrowband	
DVI4 HD	
DVI4 Ultra HD	
Linear PCM	
Linear PCM HD	
Linear PCM Ultra HD	
Linear PCM (little endian)	
Linear PCM HD (little endian)	

This section allows you to choose the codecs you prefer to use. In the list on the left, you have all the codecs that our application supports, and in the list on the right, all the codecs that you want to use and arranged in order of preference.

To use a codec from the list on the left, click on it then click "Enable >>". Conversely, if you want to remove a codec from those used, click on the one chosen in the list on the right and click on "Disable <<".

**Please note, it is recommended not to touch this section if you do not know exactly what it is and what you are doing. The default choices are intended to work in the majority of cases. So, you normally don't have to touch it.**

## 2.3 Network

This page allows you to configure the station on your network. You will be able to configure the IP address (static, DHCP or PPPoE), the server STUN, or even the RTP port.

The screenshot shows a web interface for network configuration. At the top, there is a navigation bar with 'Accounts', 'Audio', 'Network', 'System', 'Management', 'Advanced', and 'Directory'. The 'Network' section is active. On the right, there are 'Renew' and 'Logout' buttons. Below the navigation, the main content area is titled 'Network' and 'Network settings for the device'. On the right side, there is a 'Status' sidebar with a yellow warning box 'Time until auto-logout: 08:25'. The status sidebar shows 'Chef1' user information and system details: IP: 192.168.1.100 (Static), MAC Address: 18-39-19-40-05-16, System time: 2017-07-26 11:56:48, and Uptime: 21h 18m 2s. The main configuration area is divided into sections: 'WAN' (Outgoing network settings), 'STUN' (Global STUN server settings), and 'RTP' (Configure Port Range). The 'WAN' section includes 'General' (Host: InstaVoIP, Domain: InstaVoIP), 'Connection Type' (Static IP selected), 'Static IP Address' (Address: 192.168.1.100, Mask: 255.255.0.0, Default Router: 192.168.1.1, DNS Primary: 212.27.40.240, DNS Secondary: 212.27.40.241, DNS Tertiary: 0.0.0.0), 'Additional Settings' (MTU Size: 1500), and 'IPv6 is enabled!'. The 'STUN' section has 'Server' and 'Port' (3478) fields. The 'RTP' section has 'Port Range Start' (23456) and 'to' (23556) fields.

### a. WAN section

The WAN section allows you to configure your IP address and several features of your WAN interface.

The Host and Domain fields, for most users, are not to be filled out. Host is the system host name that is reported to applications and services that require it, such as DHCP or DNS. Domain is the DNS domain to which the application belongs.

Then the "Connection Type" part is more important. This is where you will choose if you want your station to have a dynamic IP address, provided by your DHCP, or if you want its address to be fixed or if you want it to be managed in PPPoE:

- **Dynamic IP Address:** If you choose this option, you have absolutely nothing to configure.
- **Static IP Address:** Here, you will need to configure:
  - o Address, which is the IP address you want to assign to your station.
  - o Mask, which subnet mask for the previously chosen address.
  - o Default Router, which is the default gateway, or the first router to which your station is connected.
  - o DNS Primary/Secondary/Tertiary, which is the IP address of one or more of your DNS servers.

### b. STUN Section

A STUN server allows a host to discover its public IP address if it is behind a NAT. This may be necessary for SIP or RTP to work properly.

Each VoIP account can be individually configured to use or not use a STUN server. This is done from that account's settings on the Accounts page. However, the system only requires one STUN server to configure, which is done here.

To use a STUN server, enter its IP address in the "Server" field, and the port used in "Port", but normally you will not need to touch this section.

### c. RTP Section

The port range chosen for inbound RTP is largely arbitrary and most users do not need to change the default settings.

Otherwise, contact your network expert.

## 2.4 System

The System page contains general administrative settings.

The screenshot shows the 'System' configuration page. At the top, there is a navigation bar with 'System' selected. Below the navigation bar, the page title is 'System' with the subtitle 'Various system specific settings'. The page is divided into three main sections: 'Authentication', 'Syslog', and 'Date & Time'. The 'Authentication' section has a 'Username' field with 'admin' and a 'Change Password' button. The 'Syslog' section has a 'Report to server' checkbox, 'Server' and 'Port' fields (with '514' in the port field). The 'Date & Time' section has 'Displayed on screen', 'Enabled', and 'Daylight savings' checkboxes, a 'Server' field with 'ch.pool.ntp.org', and a 'Time Zone' dropdown menu. A 'Status' sidebar on the right shows system information like IP, MAC, and uptime. The footer includes the 'LE LAS' logo and copyright information.

### a. Authentication

This section allows you to modify the username and password that you are asked to enter the station's web server.

### b. Syslog

This section allows you to send all log messages to a syslog server which can display or archive them. To use this function, check the “Report to server” box, indicate the IP address of the syslog server in “Server”, as well as the port used.

### c. Date & Time

This section allows you to set the time and date of your post, as well as whether it is displayed or not.

“Displayed on screen” allows you to indicate whether you want the time and date to be displayed on the LCD screen of your station, when it is idle. Check the box if you want them to be displayed. Conversely, uncheck the box if you do not want them to be displayed.



It is possible to provide an NTP server at your station. This will allow you to retrieve the time and date automatically on your network. To do this, check the “Enabled” box, and indicate the NTP server chosen (the default one works very well). Finally, indicate your time zone in “Time Zone”, as well as whether you are in summer time or not by checking the “Daylight savings” box or not.

If you don't want to use an NTP server, you can also set the time manually. To do this, click on the button provided for this purpose (“Set date & Time”), and configure the time and date as you wish.

## 2.5 Management

This page allows you to download the file containing all the station parameters, and/or to install another one. It also allows you to update your workstation if a new version exists.

The screenshot shows a web interface for system management. At the top, there is a navigation bar with tabs: Accounts, Audio, Network, System, Management (selected), Advanced, and Directory. On the right of the navigation bar are 'Renew' and 'Logout' buttons. Below the navigation bar, the main content area is titled 'System' with the subtitle 'Various system specific settings'. There are three main sections: 'Configuration' (System configuration), 'Firmware' (System update), and a 'Status' sidebar on the right. The 'Configuration' section has two sub-sections: 'Backup/Download' with a 'Download configuration' button, and 'Restore/Upload' with a file selection input and a 'Restore configuration' button. The 'Firmware' section shows the current 'Version: 4.4.4.20160603' and a 'New firmware' section with a file selection input and an 'Update firmware' button. The 'Status' sidebar on the right displays system information: 'Time until auto-logout: 09:53', 'Status: Chef1', 'User: 101@192.168.1.23', 'Status: Registered', 'System IP: 192.168.1.100 (Static)', 'MAC Address: 18-39-19-40-05-16', 'System time: 2017-07-26 14:57:08', and 'Uptime: 1d 18m 24s'.

### a. Configuration

This section allows you to retrieve the file storing all the station configurations, and to add another one.

- Backup/Download: By clicking on this button, you will download the existing configuration file in the workstation.
- Restore/Upload: Use this upload to add and replace the existing configuration file in the workstation. Be careful to add the right file type with the right structure and punctuation.

This section can allow you to configure a station, download the file resulting from this configuration and upload it to the other stations on your network so that they all have the same configuration.

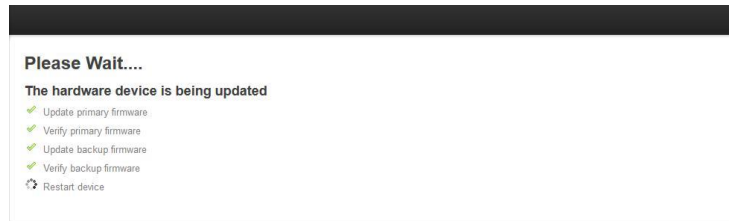
Be careful, however, to change the account information and the IP address, if static. Two stations on the same network cannot have the same IP address or the same SIP account.

## b. Firmware

This section simply allows you to update your station with new firmware. You will then select the .bin file that we sent you from your computer, using the “Browse” button, then click on “Update firmware” to apply it and update your computer.

Once the new firmware is executed, the station will verify that it has been loaded successfully, it will then make a backup copy of the new firmware and restart.

This process normally takes about 1 minute.



## 2.6 Advanced

This page allows you to fully configure the operation of your Le Las station.

The screenshot shows the 'Advanced' settings page in the Le Las web interface. The top navigation bar includes 'Accounts', 'Audio', 'Network', 'System', 'Management', 'Advanced', and 'Directory'. A 'Renew' button and a 'Logout' button are visible in the top right corner. A yellow notification bar indicates 'Time until auto-logout: 09:46'.

**Advanced**  
Choose preferred settings.

**Settings**

- Keyboard:
- Extension Keyboard:
- Allow General Call:
- General Call Number:
- Allow Conference:
- Conference Number:
- Allow Transfer:
- Transfer to:
- Priority calls:
- Push to talk:
- Call time (min):
- Ring time (s):
- Group Call:
- Micro trigger (0-15):
- AGC Micro:
- AEC:
- Auto Half Duplex:
- Buttons place:

**Relay 1**

- Password(0<->9999):
- Closing Time (s):
- Relay's options:  During Ring,  Follow Ligne,  Incoming GC,  Deactivable,  Time Gache

**Relay 2**

- Password(0<->9999):
- Closing Time (s):
- Relay's options:  During Ring,  Follow Ligne,  Incoming GC,  Deactivable,  Time Gache

**Status**  
Time until auto-logout: 09:46

**1000**  
User: 1000@192.168.1.36  
Status: Registered

**System**  
IP: 192.168.1.120 (Static)  
MAC Address: 18-39-19-41-C1-8C  
System time: 2011-10-01 08:27:29  
Uptime: 1h 40m 49s

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- **Keyboard:**  
By checking this option, you indicate that you have a complete chef station which has a complete keyboard. Conversely, if this option is unchecked, it means that you have a secondary extension with call memories only.  
How the position works differs slightly depending on what you choose. To know the difference, see the instructions for use below.
- **Keyboard extension:**  
By checking this box, you indicate to your extension that it also contains call button extensions.  
**Please note, this option can only be validated if you have a complete keyboard station.**
- **Allow General Call:**  
By checking this box, you authorize your extension to make general calls. If you do not want your extension to be able to broadcast messages throughout your network, uncheck this box. When you check this box, the “General Call Number” field will appear. If you have an IPBX server that manages the general call itself, please enter the number associated with this function here. Otherwise, leave the field empty and your general call will then be launched “peer to peer” using RTP multicast.  
  
*Attention: For group calls, please add 1 to your AG extension number in your IPBX server for each group. Example, if you configure your IPBX with extension 6700 as general call, please configure your group 1 call with extension 6701, group 2 with 6702...*
- **Allow Conference:**  
By checking this box, you authorize your extension to start a conference. The “Conference Number” field will appear. Then enter the number associated with this function in your IPBX server.  
**Attention**, this function can only be used if your extension is registered on an IPBX that supports this functionality.
- **Allow Transfer:**  
By checking this box, you authorize your extension to be able to transfer all its calls to another extension. The “Transfer to” field will appear. Then enter the number of the default extension to which you want to transfer your calls.
- **Priority Calls:**  
This option allows you to activate, or not, the call priority described in the user manual.
- **Push to Talk:**  
Defines the operating mode of Push to talk:
  - Never: implies that push to talk can never be used
  - Activable: implies that each communication will begin in full duplex, and as soon as one of the two interlocutors presses their Push to talk button, your extension will switch to half-duplex communication.
  - Always: implies that the station will operate exclusively in half-duplex.

- **Call time (min):**  
From 0 to 99: indicate here the time in minutes authorizing your extension to remain in the same communication. By putting 0 in this field, you will authorize the station to stay in communication as long as it wants.
- **Ring time(s):**  
From 0 to 99: indicate here the time in seconds allowing your extension to ring on an incoming call before picking up automatically. By putting 0 in this field, your extension can ring indefinitely.
- **Multi Calls Timer(s):**  
From 0 to 99, this field is reserved for secondary extensions and their cascade call function. This is where you determine in seconds the time a remote extension has to answer its call before it automatically switches to another extension's call. By putting 0 in this parameter, you will disable this function.
- **Multi Calls Number:**  
From 0 to 9, this field goes hand in hand with the previous one and allows you to determine the number of cascaded calls you want to assign to the press of a button. By setting 0, you will disable this cascading call function.
- **Pick up timer (0.1s):**  
From 0 to 99, this parameter, reserved for secondary stations, allows you to determine in tenths of a second the time a user must press their button to initiate a call. 0 implies instant call initiation, and 99 prevents the extension from being able to initiate a call.
- **Hang up timer (0.1s):**  
From 0 to 99, this parameter, also reserved for secondary stations, allows you to determine in tenths of a second the time a user must press their button to be able to hang up. 0 implies an instant hang-up, and 99 prevents the extension from being able to hang up.
- **Group Call:**  
The Global IP network provides access to 3 group calls.  
This field therefore allows you to define the call group(s) to which the extension belongs. Enter in this field the numbers of the groups to which you want your position to belong (Example: if I fill in 13, my position will belong to groups 1 and 3).
- **Micro trigger (0-15):**  
This field allows you to set the sensitivity of your microphone. The smaller the number you indicate, the more sensitive your microphone will be and the further away you will be able to speak. Conversely, the higher the number, the closer and louder you will need to speak for your voice to come through.  
This parameter can be useful to counter the effects of feedback between two nearby stations or for the station itself if it produces a large volume of sound at the output of its speaker.

- **AGC Micro:**  
This option enables automatic microphone gain control.  
It is not recommended to enable it.
  
- **AEC (Acoustic Echo Canceler):**  
This option enables an echo suppression mechanism.  
It is recommended to enable this option.
  
- **Auto Half Duplex:**  
This option allows you to modify the operation of full duplex.  
Instead of the microphone and speaker always being functional at the same time, the system will automatically deactivate one when the other is activated, and vice versa. Communication will therefore become half-duplex, like push to talk, except that instead of pressing or releasing a button each time to control the direction of communication, the system will switch automatically and very quickly on its own. It is therefore a half-duplex system but completely hands-free.  
The big advantage of this system is that you can have high levels of communications, both reception and transmission. So, speak from further away and have a higher listening volume, all without having feedback and being completely hands-free.  
The small disadvantage of this option is that you will have to listen to yourself and that you will not be able to speak at the same time as your interlocutor. Communicators must wait until the other person has finished what they are saying before speaking. So, you'll have to take turns speaking.  
This system must necessarily be used with type 250 stations.
  
- **Buttons place:**  
This field allows you to associate the soft keys with the physical keys of your extension.  
It is strongly recommended not to touch it without contacting us and explaining the reason for its modification!
  
- **Password Lock (0<->9999):**  
This field allows you to define a lock activation code, from 0 to 9999, that a remote station must send you to activate it.
  
- **Closing Time(s):**  
If you have chosen the lock activation option by receiving a code ("Nothing"), this parameter allows you to define the activation time before deactivating. Be careful, 0 means that the striker will not close!
  
- **Options for Waste:**  
Here you will be able to choose all the ways you want for your strike to activate:
  - **During Ring:** By checking this option, your striker will activate the entire time your extension rings.
  - **Follow Line:** By checking this option, your striker will activate all the time a communication is established.

- **Incoming GC:** By checking this option, your striker will be activated each time you receive a general call, throughout the AGM.
- **Can be deactivated:** By checking this option, each time the striker is activated, you can deactivate it by pressing one of the keys included in 0 and 9.
- **Time Waste:** By checking this option, you authorize your striker to activate upon receipt of your striker code, for the time defined in the “Striker Time” parameter.

## 2.7 Directory

It is on this page that you will be able to complete your directory.

For a manager station, apart from its classic use, the directory is also used for speed dialing with 1 and 2 digits, for example, by dialing 01 or simply 1, the station will call contact 1 in the directory.

For secondary extensions: button 1 of the extension corresponds to contact 1 of the directory...

Please note, the way to complete this differs depending on whether you use an IPBX server or not:

	Names for directory	Number using server	Number in PtoP	Which Account
Contact 1 :	Chef	100		1
Contact 2 :	186		186@192.168.1.186	2
Contact 3 :	185		185@192.168.1.185	2
Contact 4 :	Henry	125	Henry@192.168.1.26	1or2
Contact 5 :				1

As you can see, a contact has 4 fields:

- “Names for directory” contains the name with which you want your contact to appear in your directory. Whether you have a server on your network or not, it is imperative that you put something in this field.
- “Number using server” must be completed if you are likely to call this contact with an account registered on an IPBX server. In this case, this field will contain the call (extension) number saved for this contact in your IPBX server.
- “Number in PtoP” unlike the previous field, this one must be completed if you are likely to reach this contact without going through an IPBX server. Please note, this field must be completed as follows: In a single character string, without spaces, enter the identifier of the remote account (contained in the “Username/Number” field in the Account tab of the remote station), followed by of the acronym @ and followed by the IP address of the remote station.

- “Which Account” allows you to define which of the two accounts on your extension you want to call the contact with. By choosing “1”, the contact will be called with the first account of your extension (the most at the top in the “Accounts” tab if there are 2). By choosing “2”, the contact will be called with the second account of your extension (the lowest in the “Accounts” tab). By choosing “1or2”, the contact will initially be called with the first account of your extension, but if the extension detects an anomaly when sending the call, it will then cancel the call directly and call instantly with the second account of your position. Conversely if you choose “2or1”.  
This feature can be useful for a network with a main IPBX server and a second backup, in case the first fails.

### 3. GrandStream PBX Server Programming

- 1) Configure the extension ringing time to 99 seconds before being returned to the voicemail box:  
“PBX” tab --> “Internal options” menu --> “General” submenu --> “Ring Delay” field
- 2) Modification of extensions to be able to have a numbering plan between 100 and 900:
  - PBX tab -> “Internal options” menu -> “Function codes” submenu:
    - Put 1701-1720 in “Extensions for parked calls. »
    - Put 1300 in “Parked call timeout (in secs). »
    - Put 1700 in “Extension to park”
    - To safeguard
  - PBX tab -> “Internal options” menu -> “General” submenu:
    - Put 100 - 1000 in “User extension”
    - Put 400 – 499 in “Obtaining extension”
    - Put 500 – 629 in “Auto Supply.” Extensions »
    - To safeguard



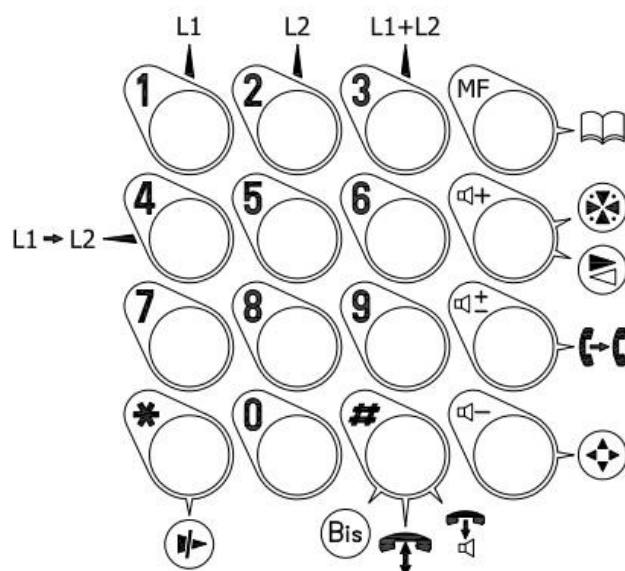


## - INSTRUCTIONS FOR USE -

### 4. Basic use of a chef station

We call head station all the stations having a display and a full keyboard.




The layout of the keyboard keys is as follows:



#### 4.1 Classic communications

##### a. Making a call

###### - With a PBX server:

- **3-digit numbering:** Using the numeric keypad, dial the 3-digit number of the remote station you want to call, and press #.
- **1-digit numbering:** Dial the number corresponding to the contact number entered into your station's web server (see Installation instructions) and assigned to the person you wish to contact, then press #. Example: 1# to call contact 1, entered in your extension's server.
- **2-digit numbering:** As with 1-digit dialing, dial the number corresponding to the contact number entered into your station's web server and assigned to the person you wish to contact. For contacts less than 10, started with a 0. Example: 01# to call contact 1, entered in your extension's server.
- **Dialing from your extension directory:** If this is entered, press the key  to open it. Then, scroll through it using the keys  Or  until you land on the contact you want to call. Finally, press the # key to initiate the call.

- **Peer to Peer mode:**

This mode has the same types of call origination as with a PBX server, except 3-digit dialing.

**b. Receiving a call**

When you receive a call, a ringtone will sound, and the name of the extension calling you will be displayed on the LCD screen. To answer this call, there are two solutions:

- Press the key #.
- let your extension pick up automatically using the parameter provided for this purpose (see Installation instructions).

**c. End of communication**


A traditional communication can end in three different ways:

- The remote station hangs up, and your station will automatically return to idle.
- Pressing the \* key on your extension will hang up.
- Your communication time, entered in the parameter provided for this purpose, has expired (see Installation instructions), the set will therefore hang up automatically.


## 4.2 General call and group call

If you have a PBX server, the definition of groups is done directly in it. In the case of a Peer-to-Peer network, this is done following the completion of the directory. As for the general call, we use the same principle, that is to say that an AG is a group call including all the stations in the network. To correctly configure your group call and general call network, refer to the installation instructions.

**a. Sending a general call**

To issue an AG, press the key  and keep the button pressed throughout your speech. When the button is released, the GA will end, and all stations will return to idle. You can issue an AG when your station is idle, or in communication. In this second case, your communication will be suspended for the duration of the issuance of your AG, and will resume directly at the end of it. General call reception has priority over all types of communication and on all extensions. If you are using a PBX server, please enter the number corresponding to the AG in your extension's web server.

**b. Making a group call**

To make a group call, enter the number of the group you want to call, then press and hold  pressed. A group call has the same operation as a general call with the difference that it only concerns the extensions in the group. However, a group call can only be initiated from an idle state of your extension, and you must absolutely be part of this group to be able to broadcast your message.

### c. AGM Reception or Group Call

If you receive a general call, or a group call concerning you, your phone will display the name of the AG transmitter, and will pick up automatically without ringing. You cannot speak to respond in this mode, it is a broadcast mode, not a call. In addition, it has priority over any other call, so if you are already in communication with an extension, this will be interrupted during the AGM, and will resume automatically at the end.



## 4.3 Conference


Conference is a mode allowing a large number of stations to speak together.

To enter the conference, press the key . When you want to leave it, press the key .

## 4.4 Carryover

When your extension is idle, there are two different ways to have all your calls transferred to another extension:

- Simply press the button  of your keyboard. All your incoming calls will then be diverted to the extension whose number is entered in the field provided for this purpose, in your web server (see Installation instructions).
- Dial the number of the extension to which you want to forward your calls and press the key .



To exit this mode, press the key again  of your position.

## 4.5 Double Call

It is possible to manage two calls simultaneously on your phone, in separate ways.

### a. Entering a second call

When you are already on a call and someone else calls you, a beep will sound and the name of this new caller will be displayed alternately with the name of the person with whom you are still on a call. Example: "New Call Chef1" and "On Line Chef2" will be displayed alternately (1s each). In this situation, two solutions are available to you:

- One press of the button  will allow you to answer this second call, and will put the extension with which you were in communication on hold.
- One press of the button  will allow you to hang up on the call that was in progress. In this case, your phone will ring again and you will then have the choice of whether or not to answer the incoming call.

#### **b. Making a second call**

To make a call in parallel with your current call, simply dial the desired number and press the # key (you can also use 2-digit dialing, or your directory). In this case, your initial call will be put on hold, and your extension will automatically move to its second line. Your call is launched, you will then hear your ringback tone, then your correspondent, once communication is established.

#### **c. Double call management**

Once in consultation, you can switch from one line to another as you wish, using buttons 1 and 2 on your keypad. By pressing key 1, your extension will then switch to line 1, that is to say your initial conversation, and by pressing key 2, you will then switch to your second conversation, on line 2 of your job.

To hang up on one of your two conversations, go to the corresponding line and press \*. You will then return to classic communication, and you will be able to receive or make a second call again.

### **4.6 Mini-conference**

This functionality simply allows three stations to communicate together on the same conversation. That is to say that you will be able to speak with two remote stations, and everyone will be able to hear each other and communicate.

#### **a. Enter Mini-conference:**

To enter this mode, you will need an initiator station. The user will first have to make a double call with the two people involved in the mini-conference, then press key 3 on their keyboard. The three of you will then meet in a mini-conference and will be able to discuss.

#### **b. Mini-conference output:**

There are two different cases to exit this mode:

- The initiating extension decides to hang up, so it presses the \* key on its extension. In this case, everyone hangs up and goes back to rest.
- One of the non-initiator stations of the mini-conference hangs up. In this case the other two stations automatically switch back to classic communication.

### **4.7 Transfer**

This functionality allows you to connect two stations from a third station. To do this, once you are on a consultation call, simply press key 4 on your keyboard. The two remote stations with which you are in communication will then be connected, as for you, you will return to idle.

### **4.8 Missed calls**

It is possible that one day, your extension, instead of being in its classic resting state, that is to say with Las displayed, the time and the date, you will have missed calls. These are stipulated by Miss>.

#### **a. Origin of missed calls:**

There are two ways for your extension to have missed calls:

- Someone called you, and you didn't answer. In this case, the identity of the caller will be taken and entered in your missed call queue.
- If someone calls you while you are already on a consultation call. Your extension is only capable of handling two calls at the same time, so if you have a third person calling you, they will be automatically rejected (except with the call priority option), but their identity will be taken, and will be entered into your missed calls queue. However, it will be stipulated to you directly thanks to a display which will be added to those indicating: who you are online with and who have you put on hold.

#### b. Management of the missed call queue:

If you have missed calls, your phone will never return to the traditional idle state, that is to say with Las written on the display, as well as the time and date. Instead, the entire call queue will be displayed, until you have processed all the calls, either by deleting them using the \* key, or by calling them back. As a reminder, there are several solutions:



- Or you only have one missed call. In this case, you just need to press the # key to contact him again.
- Or you have several missed calls. In this case, the latest one will be permanently displayed on the first line of the display, and you will just have to press # to recall it. As for the others, they will be displayed with their number alternatively on the second line of your display. You can then decide to call them back directly by dialing their number. This will therefore prevent you from having to deal with all missed calls in order of arrival.

Finally, the missed call queue is not blocking. You can use your phone as usual, it will be redisplayed to you each time your extension returns to idle, until you have processed it completely.

## 5. Advanced features of a chef station





Please note, all the functions that we will see in this part must be activated or entered in the web server (see Installation instructions).

### 5.1 Push to talk



Push to talk is a functionality allowing two stations to enter basic controlled half-duplex communication. That is to say, when one of the two stations presses its "Push to talk" button (the button  of his keyboard), he will then switch to speaking, his speaker will cut off, so he will only be able to speak, and conversely, the other remote station will switch to listening, so he will only be able to hear. When the button is released, the two stations will reverse their role, which means that you must keep your finger pressed on the button  "push to talk" throughout your speech. In addition, both positions can be piloted or piloted.

Please note, this function must be activated in your station's web server. It can be used in traditional communication, consultation calls, or even conferences. However, in the latter case, there is no longer any notion of piloting. Each of the three interlocutors can decide to switch to push to talk mode or not, and will only control the microphone and loudspeaker from their station.

## 5.2 Directory

The station has a directory accessible via the key  of the keyboard. It can be used at rest, or in basic communication between two stations. Duly entered in your station's server, it allows you to call without remembering the number of the person you want to contact. Classify in alphabetical order, you can navigate through it using the keys  And . If you keep your finger pressed on one of these two keys, the contacts will scroll more quickly. Once on the desired contact, press the # key to initiate the call. To exit the directory without calling anyone, press the key again  of the keyboard.

## 5.3 Sending DTMF codes

You can send DTMFs during a call. This can be used, for example, to send the lock code, or to activate a function on your server... To do this, when you are in communication with another station, hold down the directory button () until "DTMF Enter" appears on your screen. Once displayed, release and dial your DTMF. A "1" will send the corresponding DTMF "1", a "2" will send the DTMF "2", and this for all the keys on the keyboard, except for the four "function" keys in the fourth column of the keyboard. To exit this mode, simply press the button .

## 5.4 Strike option

The waste card is an option that we offer to our customers. It is in reality two dry contacts which can allow you to activate a flash light, open a door... Each contact can be activated by typing, from a remote station, its activation code, and this during a communication. Conversely, it can be deactivated by sending a \*. There are also 5 additional strike commands that you can activate or deactivate independently in your station's server (see Installation instructions):

- First, you can set a strike activation time. This implies that after having been activated following receipt of its code, it will automatically deactivate at the end of this defined time (see Installation instructions). This time is in seconds and can be up to 99 seconds.
- Second, the strike can be activated during the station ringing.  
Ex: Our extension is called, the ringtone begins to ring, the striker will then activate, then cut off when the ringtone goes off (answered or cancellation of the call by the caller).
- Third, like the ringtone, you can activate the strike during an entire call. When communication is established, that is to say when the two remote stations have gone off-hook, the striker will be activated until the station returns to idle. Once at rest, the striker will deactivate.
- Fourth, you can decide to activate the strike on receiving an AG.  
At the start of the GA, the striker engages, then disengages at the end.

- Lately, you can give the user the possibility of disengaging an engaged striker, by pressing a key between 1 and 9.





## 5.5 Time Limitations

It is possible to set different times leading your extension to pick up or hang up automatically (see Installation instructions):

- The allowed communication time. This time begins to count down when a communication between your extension and another remote has started, that is to say when both have picked up and started talking. Once this time has elapsed, your extension will hang up automatically.
- Ringing time. This time begins when a call is received, i.e. the first ring of your extension. Once this time has elapsed, your extension will automatically pick up.

## 5.6 Speaker volume

It is possible to adjust different volumes in your 120NTH Le Las station, but to do this refer to the installation instructions. Here we are going to talk about speaker volume. You can adjust it at any time during a call, whether in a conference, on a consultation call, or during a simple basic call between two stations. However, this volume change will only be temporary, meaning it will only last until you end the conversation. At the start of each conversation, the volume of your speaker will return to the volume defined in your station's server.

To access the volume adjustment of your speaker in a call, press the key  of your extension, then increase or decrease the volume using the keys  And . To exit the volume adjustment menu, press the key  of your keyboard.

## 5.7 Call priorities

This is an option to prioritize certain people over others to contact your extension. There are nine priority levels: positions between 100 and 199 are part of the highest priority level, these are the highest priority positions, the second level includes positions between 200 and 299, and so on until last and ninth priority level including positions 900 to 999.

We know that your extension can only handle two simultaneous calls. The question of call priority therefore arises when a third call arrives. If the "Priority Calls" option is activated, your extension will only compare the priority of the third caller with that of the one you are currently speaking with and not the one you have put on hold. If the third caller therefore has priority over this one, this new caller will then automatically take his place, and the extension with which you were chatting (lower priority), will then be ejected.

Example: you are in conversation with extension 301, and you have put extension 403 on hold. If extension 202 calls you, you will then automatically communicate with extension 202, and extension 301 will be ejected (we will hang up on them). Please note, extension 403 is still on hold, so when your conversation with extension 202 is over, you will return to communication with extension 403.



## 5.8 Managing 2 SIP accounts

Your extension may have two separate accounts. To activate the second account for your station, please check the “Second account” box in the “Accounts” tab and configure it as you wish. The configurations to be made are exactly the same as for the first account. If, however, you no longer wish to use this second account, please uncheck the “Second account” box and it will be deleted. The advantage of having two accounts on the same workstation can have several uses: your workstation is connected to two separate networks, or you want to configure a backup server...

### a. Making a call

When your station is idle, you can see written on your LCD display, either “Le Las 1”, or “Le Las 2”, or “Le Las 1P”, or “Le Las 2P”. The number indicates which account you are currently on. The “P” indicates that your account is in Peer to Peer mode. Conversely, if there is no “P”, your account is registered on an IPBX server.

This information will be important if you decide to dial to call someone. If you dial while “Le Las 1” is displayed on the screen, this will imply that your number will be sent from your first account. Conversely, if it is displayed “Le Las 2”. Please note that you can only dial a call if your account is registered on a server, which means that there is no “P” displayed after the account number. Otherwise, this implies that your account is Peer to Peer, so you will have to either use the speed dial or the directory to make a call.

To switch between accounts, hold button 1 or 2 for 2 seconds. Holding key 1 will move you to account 1, and conversely, holding key 2 will move you to account 2.

The choice of account used to send a call, when you use speed dialing or the directory, is made when configuring it in the web server (see chapter “2.7 Directory” which explains the configuration of the directory).

You now know how to choose your account to make a call, but what about receiving...

### b. Receiving and handling calls

When receiving a call, you don't have to worry about account issues. Your extension will receive all calls made to both of your accounts and those in any extension state. Having an account or 2, for receiving and handling calls, is of no importance. You can have a dual call with a call from each account, and you can do exactly the same things as if you only had one account.

The only thing is that when the station returns to its idle state, it will automatically go back to the most diligent account, that is to say the one which is registered on a server. If both are, it will default to the account its last call belonged to.

## 5.9 Station reset

You can restart your workstation in two ways:

- 1) To simply restart the station, hold down the \* and # keys at the same time for 5 seconds, then release. Your station will then restart.
- 2) Hold the \* and 0 keys for 5 seconds. You will then have two options displayed. The first: if you press key 1, the station will simply restart. The second: if you press key 2, the station will restart, but it will also modify the network parameters of your station and program your station to the IP address: 192.168.1.190 and the mask 255.255.255.0. This second reset can be useful if you have made a poor network configuration of your station and you no longer know where it is.

## 6. Basic use of a secondary station

A secondary station is any station that does not have a full keyboard and only uses memory buttons.

### 6.1 Classic communication

#### a. Making a call

A secondary station does not have a keypad, so it cannot dial a number; it only uses pre-programmed call buttons. You will therefore have to fill the extension directory with the same number of contacts as you have buttons on the extension. Thus, button 1 of your extension will automatically call contact 1 entered in the extension server, button 2 will call contact 2...

To call, simply keep your finger pressed on the button corresponding to the contact you wish to reach, for the time indicated by the parameter provided for this purpose (see Installation instructions).

#### b. Receiving a call

When you receive a call, a ringtone will sound, and the front LED will start flashing. To answer this call, there are two solutions:

- Simply press one of the buttons on your extension.
- Allow your extension to go off automatically using the parameter provided for this purpose (see Installation instructions).

### c. End of communication

A traditional communication can end in three different ways:

- The remote station hangs up, so your station will automatically go back to idle.
- Keep your finger pressed on one of the buttons on your station for the time indicated by the parameter provided for this purpose (see Installation instructions).
- Your communication time, entered in the parameter provided for this purpose, has expired, the station will therefore hang up automatically (see Installation instructions).

## 6.2 General call and group call

As a secondary extension, you cannot make general or group calls, however you can still receive them. If you receive one concerning you, your front LED will light up steadily, and your extension will answer automatically without ringing. You cannot speak to respond in this mode, it is a broadcast mode, not a call. In addition, a general call or group call has priority over any other call, so if you are already in communication with an extension, this will be interrupted during the AGM, and will resume automatically at the end .

## 7. Advanced features of a secondary station

Given its restricted keyboard, a secondary station cannot have all the functionalities of a station. However, some may apply to him:

- It can easily be controlled by push to talk by a head station.
- It can have a strike which will be activated by a remote station, or which will be activated on the station ringing, on an AG or even on a communication.
- It has the same two time limitations.
- Call priority option is also applicable. If your extension is in communication with a remote extension and a higher priority extension decides to call you, you will then automatically switch to this higher priority call and you will automatically put the extension on hold with who you were previously in communication with. Once communication with the remote station with the highest priority has ended, your station will then automatically switch back to your previous communication with the lower priority station.

However, it has a feature of its own called "Call Cascading". This function allows you to call a person, if they do not respond within a certain period of time defined by the associated parameter (see Installation instructions), automatically follow up with the call to another contact and so on within the limit of the call number parameter, associated and defined in the station server (see Installation instructions).

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**NOTES**

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