Linksys Voice System 9000 (LVS9000)
Demo kit user example

Manual configuration

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1 Introduction: Linksys Voice System 9000 demo

Linksys Voice System (LVS) is the solution designed and offered by Linksys to provide IP Telephony for SOHO and Small Businesses, with support for up to 16 users and providing key Telephony services desired by Very Small Businesses, such as autoattendant, shared line appearance, DID support, hosted voice mail, among others.

The purpose of LVS demo is to demonstrate installation, call services and additional features for both Small Businesses and Service Providers. Demonstration includes Call Control (i.e. telephony) features as well as configuration and applicability of LVS to a typical and complex business scenario.

This document explains Manual configuration option, using web user interfaces of SPA9000 and SPA9XX phones for setting up the system.

1.1 Requirements for demo setup

The following is required for performing a Live Demonstration

Linksys Voice System 9000

- 3 SPA941 IP phones, Firmware (FW) 4.1.9 or later
- 1 SPA9000 IP Key System FW 3.2.5 or later
- 1 SD208 Ethernet switch or similar
- 1 WRT54GS Broadband router
- (optional) SPA3000 PSTN GW, for PSTN routing and fallback demo

Additional equipment

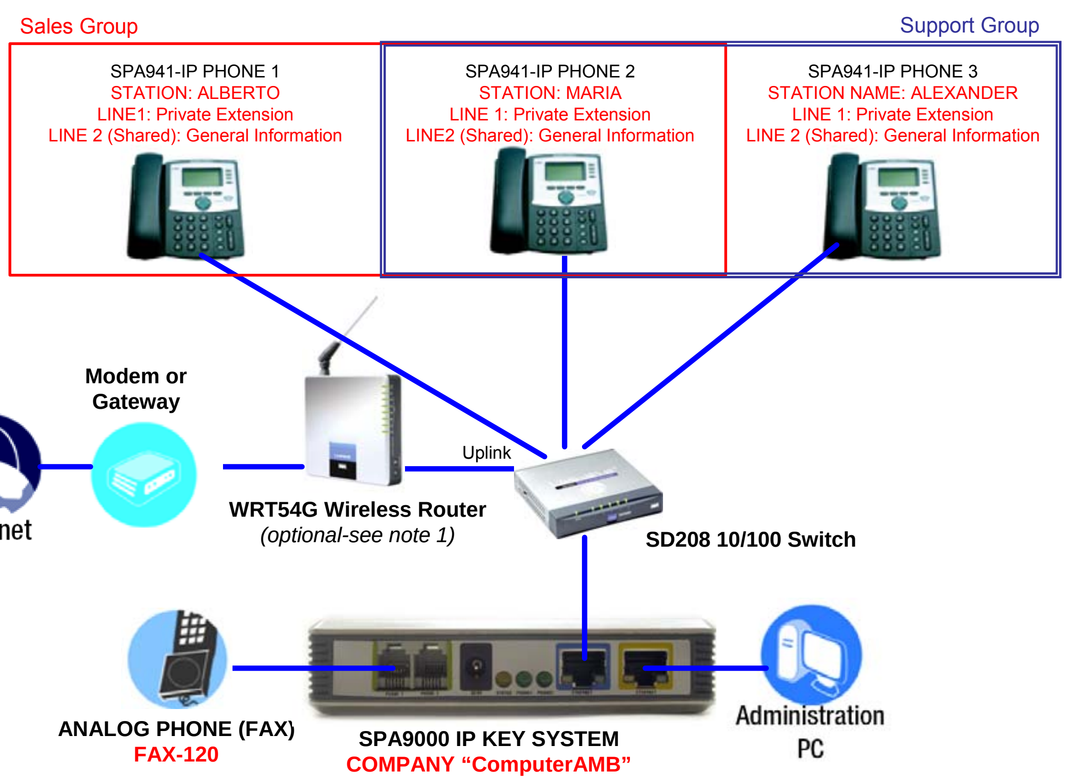
- 1 analog phone with phone (RJ-11) cable
- 1 Personal Computer with Ethernet card
- An active Internet Service connection 2 IP Telephony service accounts in order to make/receive external calls. Please consult with your local Service Providers in order to obtain two test accounts for these demonstrations.
- Internet access, allowing SIP, UDP and RTP messages to go to/from the ITSP.

2 Configuring the demo

2.1 Demo setup

Following is the setup intended for the demo. Please note this is an example only for demo purposes. The system is fully flexible to accommodate other configuration and scenarios. Please refer to the SPA9000 user guide for more information about connectivity support:

A company called <COMPANY NAME> (through this example we will use "ComputerAMB", but you may want to use a real customer name) needs Telephony services for three users plus a fax[1]. The setup is shown in the figure below:



Note 1: In case the Gateway provides DHCP server functionality WRT54G is not necessary, i.e. the switch (uplink port) can be connected directly to the Gateway/Router.

The Company scenario is as follows. Company Computer AMB is a very small business focused on sales of Personal Computers for SMB. In addition they provide technical

[1] Please note standard SPA9000-EU provides support for 4 users which can be extended to 16 users by purchasing a license key (SKU: SPA9000UPG).

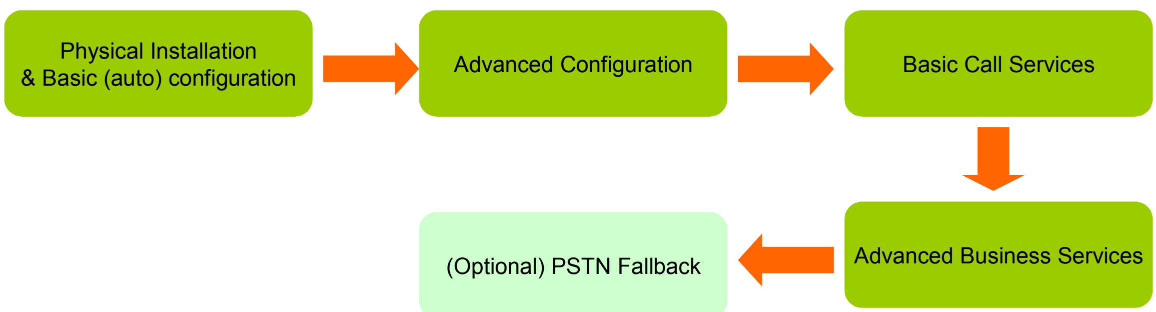
support services to their sales. As summary these are the main interactions/activities of users:

- The office has three people, i.e. ALBERTO, MARIA and ALEXANDER
- One Fax (simulated by an analog phone in the demo)
- They run two separates activities: Sales and Support
- ALBERTO and MARIA run the Sales activities
- ALEXANDER run the Support activities whilst MARIA is backup for Support
- All three handle General Inquiries
- Working hours: Monday-Friday, from 9:00 am to 6:00 pm, Saturday-Sunday closed.
- Company location: Main Avenue 2, Main Industrial Park, first floor. Madrid 28001. Spain.

The connectivity scenario to the Internet may be a typical Residential/SOHO/SB Internet Access connection, e.g. based on ADSL.

2.2 Demo Use cases

Diagram below shows the sequential steps of the LVS demo, covering installation, configuration, basic and advanced business services and survivability.



1. Physical Installation and Basic (auto) configuration: It includes cabling, and physical connectivity of equipment. In addition, this step shows the auto configuration feature which enables the system to perform internal calls without any additional configuration (plug and play).
2. Advanced Configuration: It covers the configuration processes to enable advanced features as well as some additional characteristics such as change of screen appearance. The following features are configured at this stage:
    a. Configuration of Service Provider accounts to make external calls.
    b. Phone appearance – Configuration to customize the phone appearance, i.e. adding a user assigned station name as well as name for lines, disabling unused line keys, etc.

   c. Regional settings – Setting Day and Time[2]
   d. Shared line configuration – Enabling several users to share same line.
   e. Hunt Groups – Hunt groups are groups for receiving incoming calls which belong to the same organizational area
   f. Autoattendant – Configuration of the main autoattendant according to Business requirements, i.e. setting the welcome prompts, defining the autoattendant tree, etc.
3. Basic Call Services: A demonstration of basic call services
   a. Originating and terminating calls
   b. Call forwarding
   c. Music on Hold
   d. Personal Directory
   e. Speed Dial
   f. Call History
4. Advanced Business Services: A demonstration of advanced services for Business Users
   a. Shared line
   b. Call transfer (attended and blind)
   c. Conference
   d. Parking and unparking calls
   e. Call Pickup (directed and group)
   f. Intercom
   g. Group Paging
   h. Autoattendant
   i. Do not Disturb
5. (optional) PSTN Fallback: A demonstration of SPA3000 PSTN Gateway connectivity and configuration to enable PSTN fallback in case of failure of broadband access. It enables the Telephony system to stay online even in case of Internet access failure or failure on the Internet Service provider. For information about SPA3000/SPA3102 integration into the LVS9000, please refer to document *SPA3000/SPA3102 integration with SPA9000*.

2.3 Physical Installation and basic plug&play procedure

Follow these instructions to install the system for internal calls

1. Connect an Ethernet network cable to the Internet Port of the SPA9000. Then connect the other end of the cable to one of the local ports on your switch.

[2] Regional settings incluyes other ítems, such as supplementary services activation codes, tones setting, etc, which are not part of the demo in order to keep it as simple as possible and to focus on user features.

2. Connect an Ethernet network cable to the Ethernet Port of the SPA9000. Then connect the other end to the computer you will use to configure the SPA9000 (this is called the administration computer).

3. Plug an analog phone into the Phone port 1 of the SPA9000.

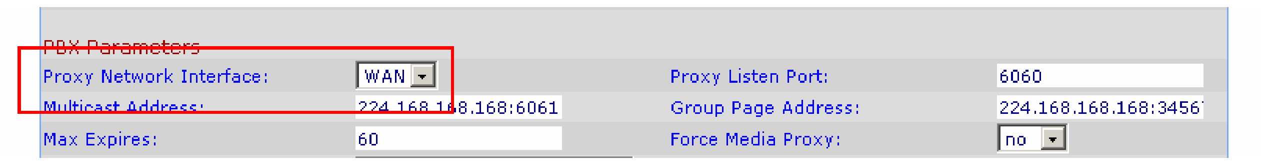
4. Connect the included Power Adapter to the power port of the SPA9000, and then plug the power adapter into an electrical outlet. The status LED will start flashing as the System boots up.

5. (*Optional – Only necessary in case Customer Internet access uses a Modem or Router/GW without DHCP server option*) Install the WRT54GS Wireless Router.
    a. Connect an Ethernet network cable from the Customer LAN modem/router to the Internet port of WRT54GS
    b. Connect an Ethernet network cable from one of the LAN ports of WRT54GS to the Uplink port of SD208. *Note that in case WRT54GS is not used, Uplink port of SD208 must be connected to a LAN port of Customer Router/GW*.
    c. Connect the included Power Adapters to the power ports of WRT54GS and SD208, and then plug the power adapters into the electrical outlets.
6. Connect an Ethernet network cable to the Uplink port of the switch. Then connect the other end of the cable to one of the ports of the WRT54G Wireless Router or to the Customer Router Gateway.

*Note: It is recommended to reset SPA9000 before starting configuration: Connect the analog phone to FXS port 1, then press **** and 73738# and then 1#. Wait for 30 seconds.*

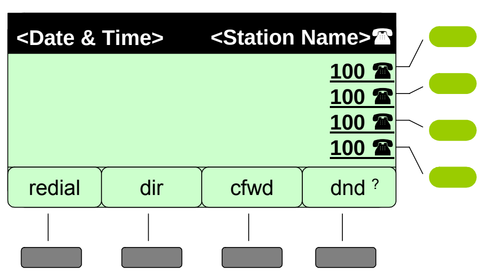
7. From the Administration Computer, launch an Internet browser application (Internet Explorer, Netscape Navigator, or similar)
8. Enter http://192.168.0.1/admin/voice/advanced in the address Field (**192.168.0.1** is the default local IP address of the System). Then press the **Enter** Key. The *Voice-Info* screen will appear.
9. Make sure the proxy interface is set to WAN
    a. Click SIP Tab
    b. Under PBX parameters, check Proxy Network Interface parameter is set to WAN as shown in figure below

c. In case Proxy Network Interface parameter is set to LAN, change it to WAN
d. Click Submit All Changes button. SPA9000 will reboot. Wait for 30 seconds.



10. Connect an Ethernet network cable to an IP Phone (SPA941). Then connect the other end to one of the ports in the SD208 switch.
11. Connect the included power adapter to the power port of the IP phone, and then plug the power adapter into an electrical outlet.
    a. If the IP phone has been used before, reset it to its factory default settings first, by pressing Setup button (🗎) and option 14 Factory Reset, then press select.
12. The IP phone will reboot 2-3 times (the rebooting may take up to one minute). The system will automatically assign an extension number to the IP phone. When the IP phone displays it extension number, then it is ready for use.
13. Repeat steps 7-9 untill you have installed all your IP phones (this demo includes 3 phones, but you can plug up to 4 SPA94X in this basic configuration)

Following is the aspect of the screen after basic autoconfiguration is complete. All line keys are directed to the extension line (e.g. 100[3,4]).



In the advanced configuration, you will modify the appearance of the phone screen by configuring station name, setting Date and Time and configuring line keys.

Congratulations! You have completed the basic installation of the demo. You should be able to place internal calls. As you can see, basic configuration is performed automatically in the system!

[3] Please note that extension number may change depending on whether the SPA9000 has been used before.
[4] This extension numbers are assigned automatically, if you prefer to use a different extension number convention, please modify the User ID in the respective IP Phone “Line” tab or modify the Next Auto User ID parameter in the SIP tab of SPA9000.

2.4 Advanced Configuration

2.4.1 Configuration for External calls

Before you proceed please make sure you have an Active Internet connection and have setup two accounts with an IP Telephony Service Provider. For the purpose of the Demo please identify one account as Company Main number and second account as fax number.

For each accounts you must have at least[5] following information available:

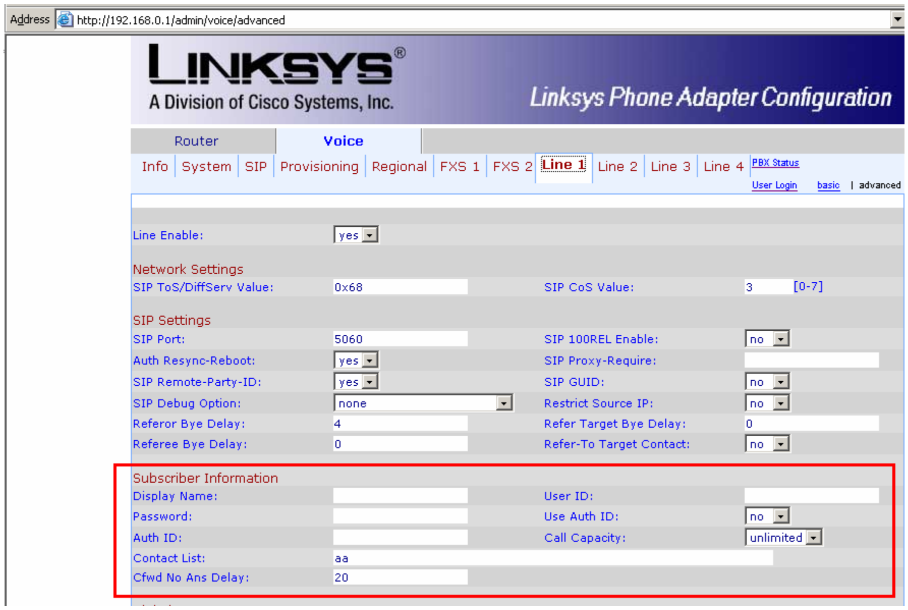
Parameter name	Type	Description	Location in Web User Interface
SIP Proxy address	URL (example: sip.foo.com)	This is the address of the SIP Server where SPA9000 will connect to	<Line "n"> tab, being n from 1 to 4
Outbound SIP Proxy address (optional)	URL	This is the address of the SIP server where request if first sent to. This is optional.	<Line "n"> tab, being n from 1 to 4
User ID	String (example:3491700398098)	This is the user identifier. It is typically mapped to the user telephone number	<Line "n"> tab, being n from 1 to 4
Password	String	This is the password for the user account	<Line "n"> tab, being n from 1 to 4
Authentication ID (optional)	String	This is an additional identifier used for authentication purposes. It is optional. If available, the "Use Auth ID" parameter must be set to Yes	<Line "n"> tab, being n from 1 to 4

1. From the Administration Computer, launch an Internet browser application (Internet Explorer, Netscape Navigator, or similar)
2. Enter http://192.168.0.1/admin/voice/advanced in the address Field (**192.168.0.1** is the default local IP address of the System). Then press the **Enter** Key. The *Voice-Info* screen will appear.
3. Click the Line 1 tab.
4. On the Line 1 screen, enter the Line settings as described below

2.4.1.1 Subscriber Information

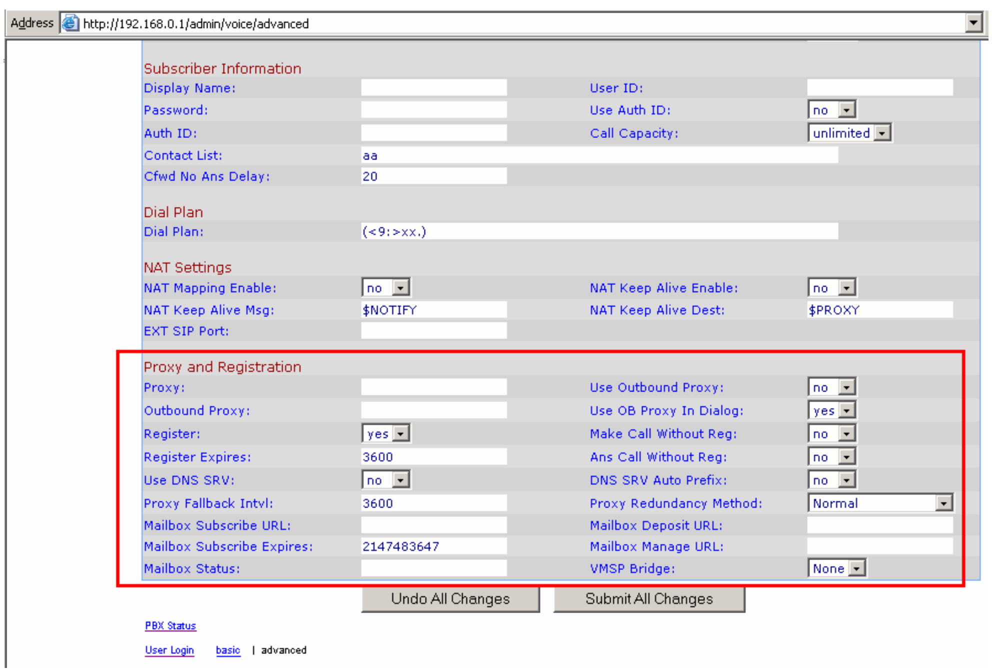
- User ID, also called account number, supplied by your Internet Phone service provider. Do not use hyphens or spaces.
- Password, enter the case-sensitive password supplied by your Internet Phone service provider.
- Display Name: This is the name that will appear on called screen when you perform a call over the line.
  - In this demo, use CompanyAMB MAIN for Line 1 and CompanyAMB Fax for Line 2[6]

[5] Note some Service Providers may require configuring more parameters than those indicated in this demo setup. The parameter list shown here includes the most common parameters used by Service Providers.
[6] Third line will be used for PSTN fallback and fourth line is not used in this demo. A typical example is the use of an exclusive support line directed to the support group (in this demo the support group is reachable through the auto-attendant).



2.4.1.2 Proxy and registration

- Proxy. Enter the proxy address supplied by your Internet Phone service provider.



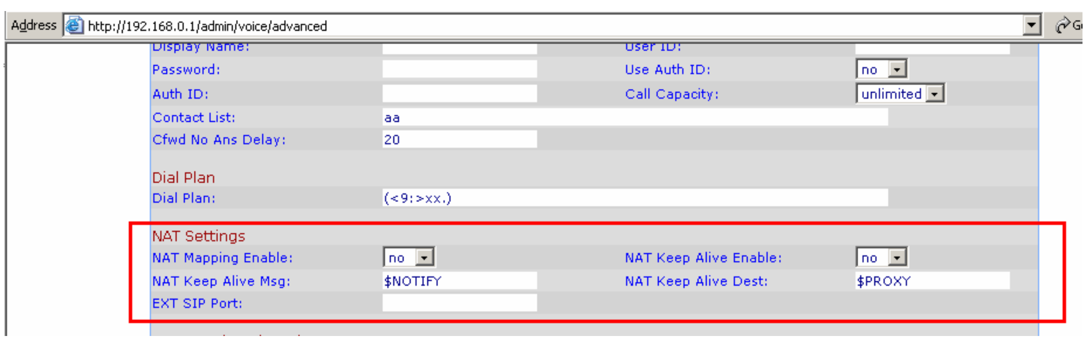
2.4.1.3 Music On Hold for Parked calls

- Click SIP Tab and Check *Call Parking MOH Server* is set to imusic, as shown in figure below.

2.4.1.4 NAT settings (optional)

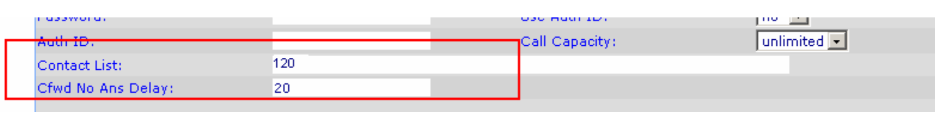
NAT settings are necessary in some cases where there are several NAT functions across the path (e.g. using WRT54G and an additional router, both doing NAT) that may have different time settings for port mapping lifetime. In this case set the following parameters:

- NAT Keep Alive Enable to Yes



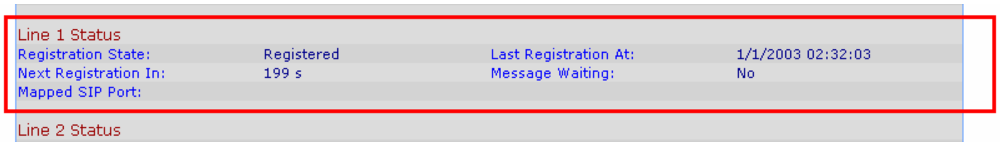
If your Internet phone service provider supplied additional settings (e.g. SIP port, authentication ID, outbound SIP proxy address), make sure you enter those as well. Refer to the instructions your Internet phone service provider gave you.

5. Repeat (3-4) for Line 2 (fax)
6. For Line 2, set the Contact List to Fax line (120). Second line is used as a Fax DID (any caller dialing Line 2 external number will be routed directly to Fax)



7. Click the **Submit All Changes** button to save your new settings.
8. The system will reboot itself and then the IP phones will reboot themselves.

At this time you should be able to make and receive external calls (access external line with “9”). In order to ensure you are registered check the **Voice-Info** tab in http://192.168.0.1/admin/voice/advanced



At this point, you can make and receive calls with the IP phones!

2.4.2 Enabling Analog phone/fax

In order to enable origination and reception on the Analog phone please follow following steps:

1. Enter http://192.168.0.1/admin/voice/advanced in the address Field (**192.168.0.1** is the default local IP address of the System). Then press the **Enter** Key. The *Voice-Info* screen will appear.
2. Click the FXS 1 tab.
3. On the FXS 1 screen, enter the settings

2.4.2.1 Subscriber Information

- User ID, also called account number, supplied by your Internet Phone service provider. Do not use hyphens or spaces. In this example we use *120 as FAX extension number.*
- Display Name; enter the case-sensitive password supplied by your Internet Phone service provider. In this example we use ComputerAMB Fax

- Click the **Submit All Changes** button to save your new settings.

Congratulations! You should be able to place internal and external calls (dial 9), and also be available to receive calls using the standard autoattendant for MAIN line and incoming calls over the second line routed to the fax. In next cases we will use more advances features for customizing even more the Linksys Voice System.

2.4.3 Appearance, Regional Settings and Shared Line appearance

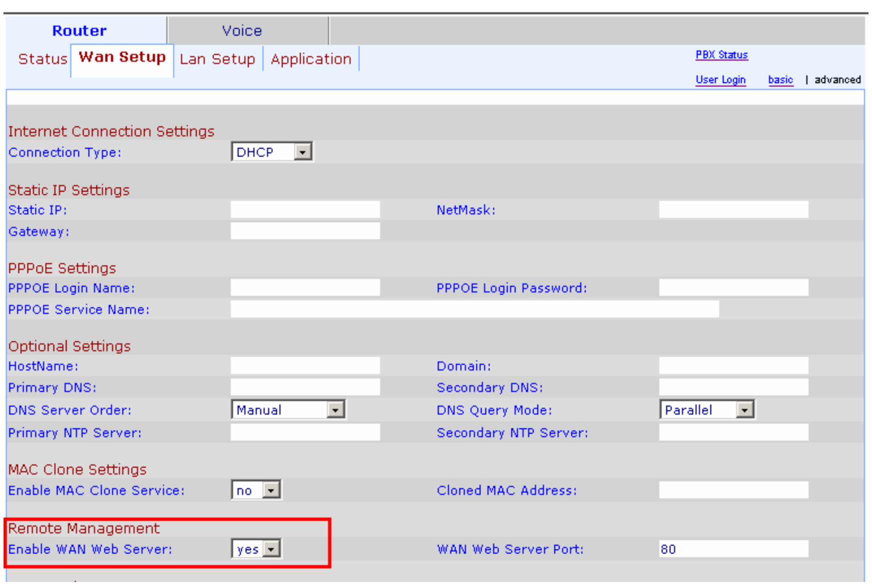
In this step, we will configure the following:

- Phone screen appearance, including station name, disabling unused line keys
- Regional settings: Date and Time using NTP server
- Shared line: We will configure a shared line for General Information services. This line will be shared among all users, and will ring when any caller requires General Information. We will use extension number 200 for it.

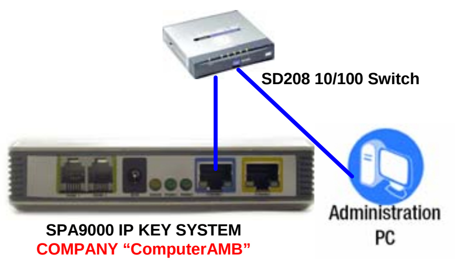
Note this procedure is to be repeated for each phone.

Before starting customization of Phone screen, enable WAN access server in SPA9000, This will enable to connect the administration PC to the switch so you will be able to configure both SPA9000 and IP phone with the admin PC.

a. Enter http://192.168.0.1/admin/advanced in the address Field (**192.168.0.1** is the default local IP address of the System). Then press the **Enter** Key. The *Router-Status* screen will appear.
b. Select Wan Setup tab
c. Set **Enable WAN Server** parameter to Yes



d. Click **Submit All Changes**
e. Move the admin PC cable from the SPA9000 Ethernet port (Yellow port) to a switch port.



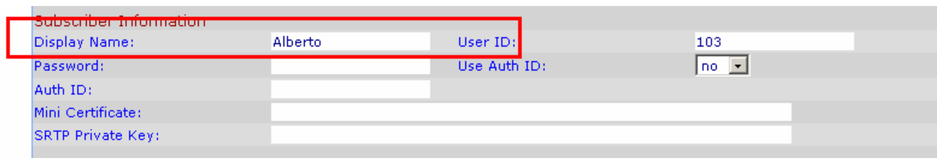
f. From now on, the SPA9000 Administration User Interface is located at, http://<SPA9000_WAN_address>/admin/advanced. In order to know the WAN IP address, from the analog phone press ********, then **110#**, and you will hear the WAN IP address.

2. Configure first phone (Alberto)

2.4.3.1 Phone Display appearance

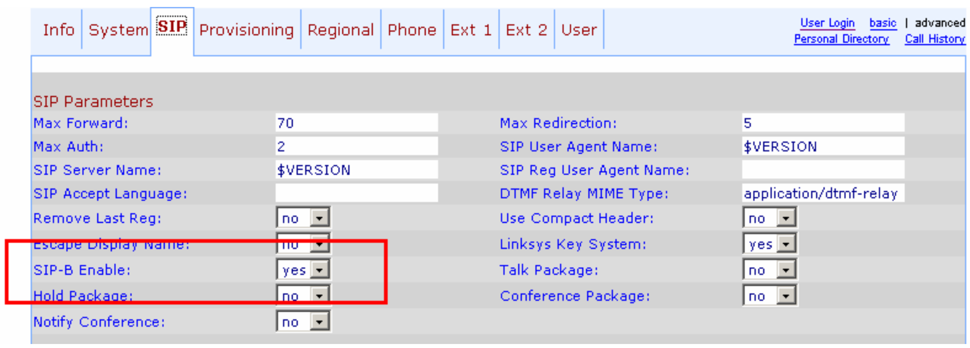
a. Connect to the first phone (ALBERTO station)
b. Identify the IP Phone IP address, by pressing Setup button (🗎) and option 9.
c. Open an Internet browser window and type following address: http://<IP_Phone_IP_address/admin/advanced, the IP phone Web Administration page will open.
d. Configuration of Station, and Line Names
    i. Click on the **Phone** tab
    ii. Change **Station Name** parameter to ALBERTO
    iii. Change Line Key 2 **Extension** (currently 1) to **2**
    iv. Change Line Key 2 Shared Call Appearance to **shared**
    v. Change Line Key 2 Short Name (currently 10X) to **ComputerAMB**
    vi. Change Line Key 3 and Line Key 4 **Extension** (currently 1) to **disable**

vii. Click on **Ext 1** tab
viii. Set **Display Name** to Alberto

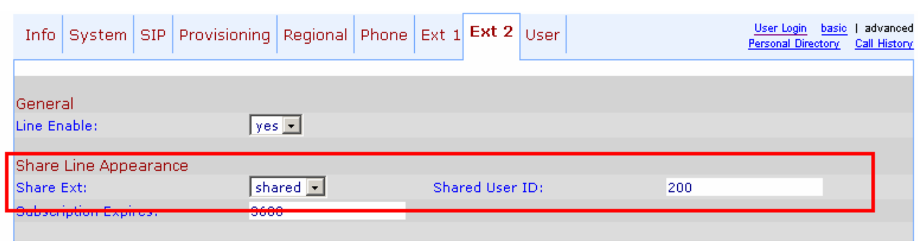


2.4.3.2 Configuration of Shared Line

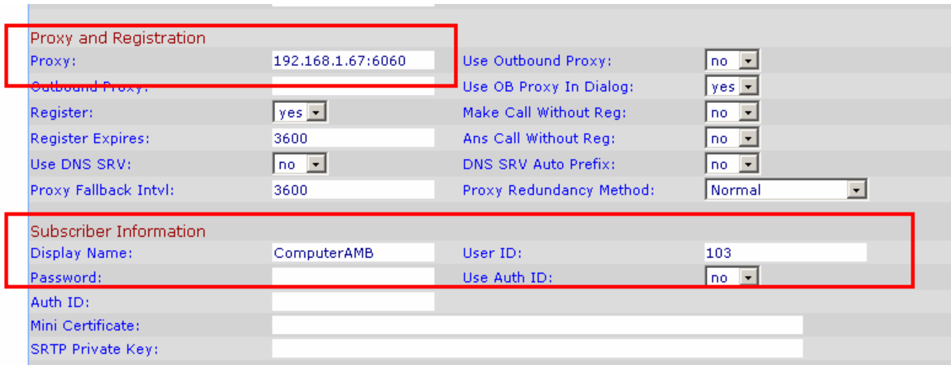
i. Click on the SIP tab
ii. Set **SIP-B enable** parameter to Yes



iii. Click on the **Ext 2** tab
iv. Set **Share Ext** parameter to **shared**
v. Set **Shared User ID** parameter to **200**

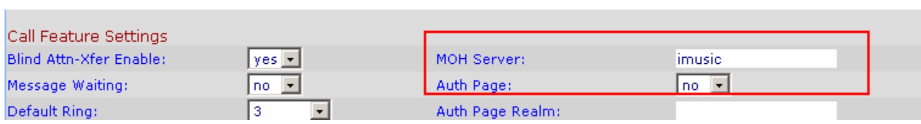


vi. Set **Proxy** parameter to the SPA9000 to the SPA9000 WAN IP Address, obtained it from step 1-f (****, then 110#), and setting port to 6060
vii. Set **Display Name** parameter to ComputerAMB
viii. Set **User ID** parameter to 10X (same number as ext 1)
ix. Click **Submit All** Changes



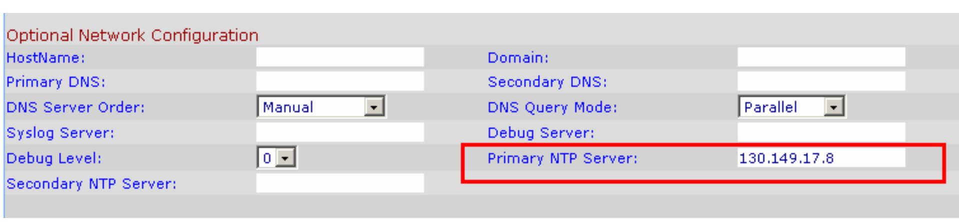
2.4.3.3 Configuration of Music On Hold

i. Click on Ext 1 tab
ii. Set MOH Server to imusic
iii. Repeat step on Ext 2 tab



2.4.3.4 Configuration of Date and Time

i. Click on the **System** tab
ii. Set **Primary NTP Server** parameter to 130.149.17.8 which correspond to public NTP server ntps1-1.cs.tu-berlin.de[7]



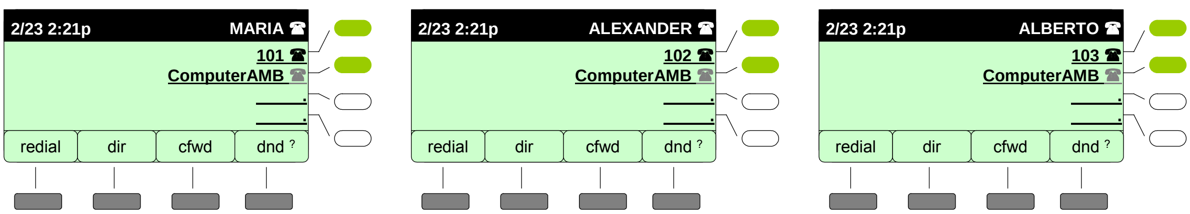
iii. Click on the **Regional** tab
iv. Set **Local Date (dd/mm)** parameter
v. Set **Local Time (HH/mm)** parameter
vi. Set **Time Zone** parameter.

[7] Check with your IP Telephony Service Provider availability to use their public NTP Server if available.

Miscellaneous			
Set Local Date (mm/dd):	02/23	Set Local Time (HH/mm):	12/51
Time Zone:	GMT+01:00	Time Offset (HH/mm):	
Daylight Saving Time Rule:	start=4/1/7;end=10/-1/7;save=1		
DTMF Playback Level:	-16	DTMF Playback Length:	.1
Inband DTMF Boost:	12dB		

3. Repeat step 2 for the remaining phones
    b. Maria
    c. Alexander

At this point, each phone is configured with a private line, and a shared line (ComputerAMB, used for general inquiries). Your phone screens should looks as below:



2.4.4 Incoming call Configuration

Auto-attendant Configuration includes configuration of voice prompts and interaction scripts. In this example we will setup the incoming call configuration:

1. Callers which are dialing main number will be attended by auto-attendant.
2. Callers which are dialing to the fax number will be directed to fax extension

2.4.4.1 Hunt Groups Configuration

As part of the auto-attendant configuration, we need first to configure the hunt groups, i.e. groups which are going to receive calls belonging to the same group. In this example, there are two groups: Sales and Support.

- Sales is formed by Alberto and María extensions (in this example 103 and 101 respectively)
- Support is lead by Alexander extension and Maria as back up (in this example 102 and 101).
  We are assigning the following hunt group extension numbers[8]:
    - Support group: 500
    - Sales group:600

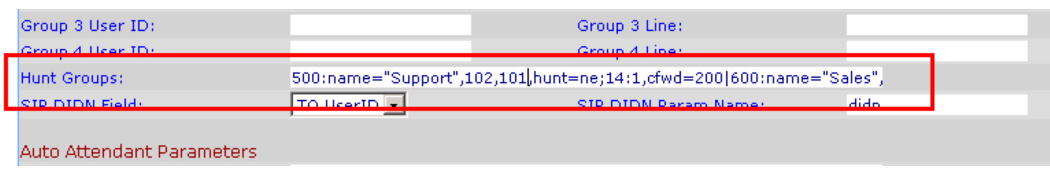
Following is the script to place in the **Hunt Groups** parameter (SIP tab):

[8] Note that any extensión in the system can dial to the hunt group (i.e. it is not only for external dialing but also applicable to internal calls).

```
500:name="Support",102,101,hunt=re;14:1,cfwd=200|600:name="Sales",103,101,hunt=
ne;14:1,cfwd=200
```

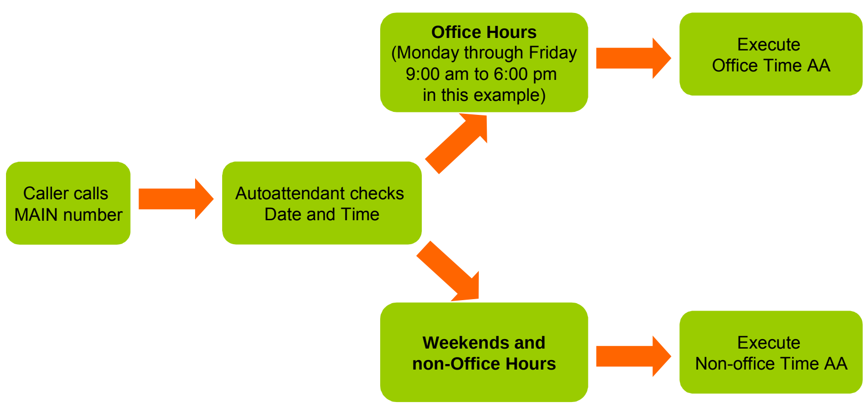
The behavior of the hunt group is the following:

- Any time someone dials 500, it is directed to support group, ringing Alexander first "always", in case it is not answered then it is fwded to María. If not available the forwarded to the main line.
- Any time someone dials 600, It is directed to Sales group, ringing either Alberto or María (depending who answered latest group call). If not available then forwarded to the main line.



2.4.4.2 Autoattendant Configuration[9]

Auto-attendant is a fully flexible and customizable feature which enables the Small Business to have a customized automatic attendant including new prompts. In this example we are going to configure two autoattendant prompts, one for the working hours and another for weekends and non-office hours which will drive the user interaction, when an external caller calls the company through its main number.



To configure the autoattendant the following needs to be configured:

1. Setting of Office hours and autoattendant usage
2. Office hours autoattendant script
3. Non-office hours autoattendant script
4. Autoattendant dial plans
5. Autoattendant voice prompts

[9] Please note this configuration is an example only. The Autoattendant is fully configurable, both in terms of prompts as well as user interaction. Please consult the SPA9000 user guide for more information.

2.4.4.2.1 Office hours and autoattendant usage (in SIP tab)

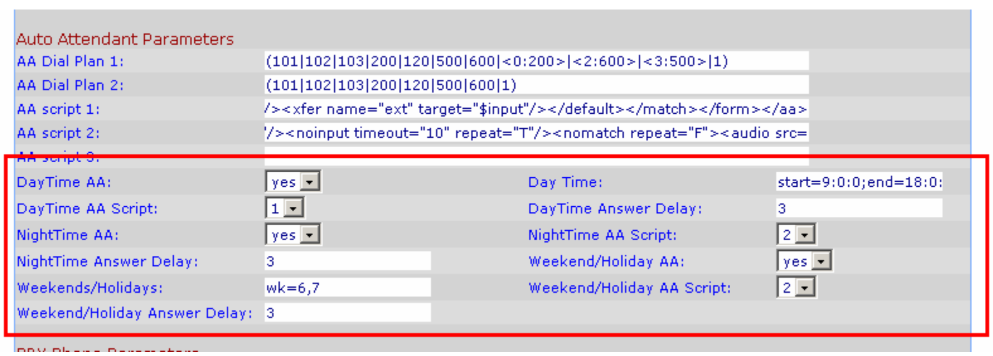
1. Set **DayTime AA** parameter to Yes
2. Set **DayTime** parameter, from 9:00 am to 6:00 pm

```
start=9:0:0;end=18:0:0
```

3. Set **DayTime AA script** parameter to 1
4. Set **DayTime answer delay** parameter to 3 (seconds)
5. Set **NightTime AA** parameter to Yes
6. Set **NightTime AA script** parameter to 2
7. Set **NightTime answer delay** parameter to 3 (seconds)
8. Set **Weekend/Holiday AA** parameter to Yes
9. Set **Weekend/Holiday** parameter to Saturday (6) and Sunday (7)

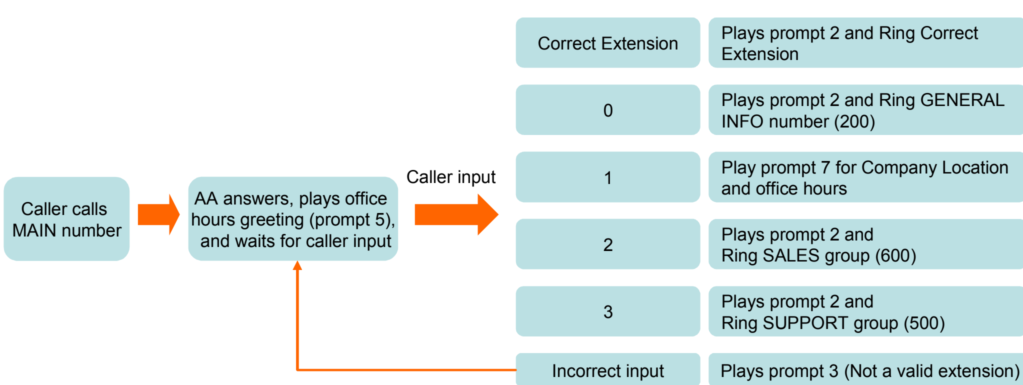
```
wk=6,7
```

10. Set **Weekend/Holiday AA** script parameter to 2
11. Set **Weekend/Holiday answer delay** parameter to 3 (seconds)



2.4.4.2.2 Office hours autoattendant script

Figure below shows the auto-autoattendant interactions for callers to the ComputerAMB main number during office hours:



a) Caller dials ComputerAMB main number during office hours
b) AA answers and plays Office Hours welcome greeting (Prompt number 5) and waits for caller DTMF input
   a. If user enters correct extension, the Autoattendant plays forward announcement (Prompt number 2) and forward the call to the extension number entered by the user
   b. If user enters "0", the Autoattendant plays forward announcement (Prompt number 2) and forward the call to the General INFO number (200 in this example)
   c. If user enters "1", the Autoattendant plays the Company Location and office hour announcement, then it goes back to main AA prompt (number 5).
   d. If user enters "2", the Autoattendant plays forward announcement (Prompt number 2) and forward the call to the SALES Group number (600 in this example)
   e. If user enters "3", the Autoattendant plays forward announcement (Prompt number 2) and forward the call to the SUPPORT Group number (500 in this example)
   f. If the user enters an incorrect extension, the Autoattendant plays an announcement indicating the extension number is not correct (prompt 3) and goes back to main AA prompt (number 5).

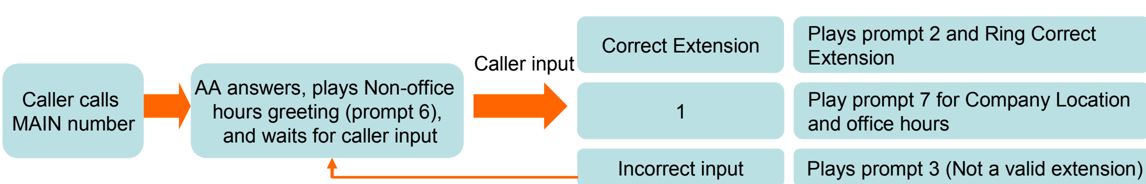
To configure office hour AA script:

- Set the **AA script 1** parameter (SIP tab) to the following script[10]:

```
<aa><form id="dt" type="menu"><audio src="prompt5" bargein="T"/><noinput
timeout="10" repeat="T"/><nomatch repeat="F"><audio src="prompt3"
bargein="T"/></nomatch><dialplan src="dp1"/><match><case input="1"><audio
src="prompt7" bargein="F"/></case><default><audio src="prompt2"/><xfer
name="ext" target="$input"/></default></match></form></aa>
```

2.4.4.2.3 Non-office hours autoattendant script

Figure below shows autoattendant interactions for callers to the ComputerAMB main number during Non-office hours:



a) Caller dials ComputerAMB main number during Non-office hours
b) AA answers and plays Non-Office Hours welcome greeting (Prompt number 6) and waits for caller DTMF input

[10] This script implements the AA behavior explained in the example. AA scripts can be configured in a different manner to implement different behavior. It is recommended to read the SPA9000 user guide for more information about AA scripts.

a. If user enters correct extension, the Autoattendant plays forward announcement (Prompt number 2) and forward the call to the extension number entered by the user.
b. If user enters “1”, the Autoattendant plays the Company Location and office hour announcement, then it goes back to main AA prompt (number 6).
c. If the user enters an incorrect extension, the Autoattendant plays an announcement indicating the extension number is not correct (prompt 3) and goes back to main AA prompt (number 6).

```
<aa><form id="dt" type="menu"><audio src="prompt6" bargein="T"/><noinput
timeout="10" repeat="T"/><nomatch repeat="F"><audio src="prompt3"
bargein="T"/></nomatch><dialplan src="dp2"/><match><case input="1"><audio
src="prompt7" bargein="F"/></case><default><audio src="prompt2"/><xfer
name="ext" target="$input"/></default></match></form></aa>
```

2.4.4.2.4 Autoattendant dial plans (in SIP tab):

Autoattendant dial plans are used for definition of the allowed user DTMF input numbers, as well as for digit substitution to handle dialing options. There is one dial plan for the office-hours AA (dial plan 1) and another one for the non-office hours and weekends (dial plan 2). To configure the dial plans follow the steps mentioned below:

a) Enter the following string in the AA dial plan 1 parameter:

```
(101|102|103|200|120|500|600|<0:200>|<2:600>|<3:500>|1)
```

This dial plan accepts the following extension numbers: 101, 102, 103, 200, 120, 500 and 600, and accept 0,1,2,3 as DTMF entries (for AA menu) performing substitution to the corresponding extension numbers[11].

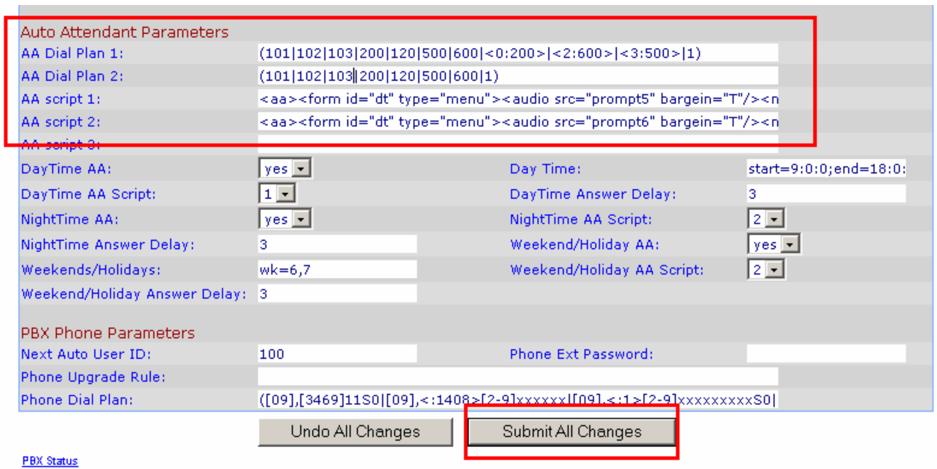
b) Enter the following string in the AA dial plan 2 parameter:

```
(101|102|103|200|120|500|600|1)
```

This dial plan accepts the following extension numbers: 101, 102, 103, 200, 120, 500, 600, and accept 1 as DTMF entry for AA menu.

c) Click **Submit All Changes** button and wait for the unit to reboot.

[11] This dial plan implements the AA behavior explained in the example. AA dial plan can be configured in a different manner to implement different behavior. It is recommended to read the SPA9000 user guide for more information about AA dial plan.



2.4.4.2.5 Autoattendant voice prompts:

The following are the voice prompts to configure, please note that you can configure it in your language.

Prompt	Description	Message
1[12]	Basic Message – Welcome	"If you know your party's extension, you may enter it now"
2	Basic Message – Forward	"Your call has been forwarded"
3	Basic Message – Not valid	"Not a valid extension, please try again"
4[12]	Basic Message – Bye	"Goodbye"
5	Office Hours message	"Thank you for calling ComputerAMB, if you know your party's extension number, you may dial it at any time. For General Information dial 0, for company location and hours dial 1, for sales department dial 2, for support dial 3".
6	Non-Office Hours message	"Than you for calling ComputerAMB, currently we are closed. For company location and office hours dial 1, if you know your party's extension number, you may dial it at any time".
7	Company location and Office Hours	"We are open Monday to Friday 9 am to 6 pm. Our office location is Main Avenue 2, Main Industrial Park, first floor. Madrid 28001".

a) Using the analog phone connected to Phone 1 port of SPA9000, enter ******** to access the IVR
b) Enter **72255#** to access autoattendant message menu
c) Select the message you would like to record and press #
d) Follow the IVR instructions to record and review the message.
e) Follow the steps for message 2, 3, 5, 6, and 7.
f) Exit the IVR (hang up).

Congratulations! You have configured the entire system. It is now time to play with the features!

[12] Messages 1 and 4 are not used in this example.

3 Demonstrations

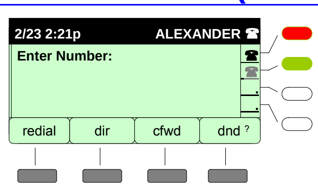
Before starting the Basic and Advanced Call Services demonstration please check the phone structure, buttons and their names as they are used through the demonstration

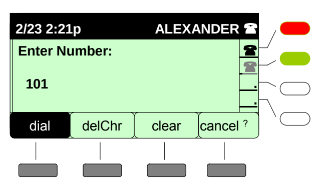
3.1 Basic call services

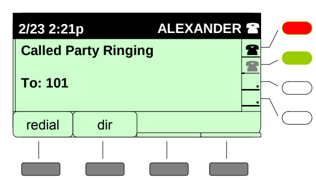
3.1.1 Basic Call

This example shows basic call establishment where ALEXANDER-102 calls MARIA-101.

ALEXANDER-102 (Caller)





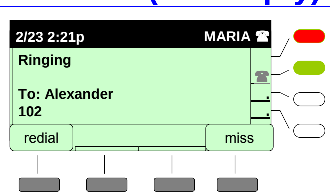


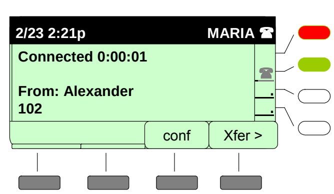
1.- Alexander: (a) Off-hook the phone or (b) press the speaker button, (c) press the line key to get access to the line, or (d) Press **dir** softkey to get access to the personal/corporate directory

2.- Alexander: Enter the number to dial (prefix 9 if you would like to reach an external number)
3.- Press “dial” Softkey, or press “#” or wait for the digit timer expiration.

4.- Alexander: Wait for the other end (María-101) to answer

MARIA-101 (Called pty)





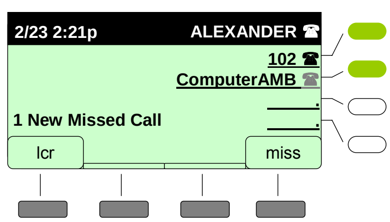
4a.- María: Phone is ringing

6.- María answer the phone: (a) Pick-up the earphone, (b) Press the speaker button or (c) press the line key

Call is established. To end the call, either party can (a) Hangup the earphone, (b) press the Line key in use

3.1.1.1 Dial options

i. *Direct dialing*: As simple as dialing the extension number. For an external call, press 9 to get outbound dial tone[13]
ii. *Via directory*: There are two directories, Personal and Corporate. Personal directory is stored locally in the phone, either manually or via the web user interface. Corporate directory includes all extensions which are subscribed to the SPA9000. In order to call via directory, press the **dir** softkey in main screen, then select the entry or **Corporate Directory**, and press **select** key. Then move the cursor to the other party (e.g. Alberto 103) and press **dial**.
iii. *Redial*: Redial is used to dial a number which was dialed in the past. Press the **redial** softkey in the main screen in order to get the redial list. It indicates the number dialed and date and time. Then navigate through the list with the navigation key to the number to (re)dial, and press **dial** key. Note it is also possible to **delete**, **edit** or **save** the entry.
iv. *Last Received Call*: Last Received Call (LCR) is used to automatically dial caller of the last received call. To dial LCR press the **lcr** softkey. Lcr softkey is on first position if there is a missed call or fifth position if there is no missed call (use the navigation key to the right to search for it).
v. *Missed Calls*: In case there is a missed call, the phone display will look as the figure below indicating the number of missed calls, allowing the used to callback. In order to dial:



Press the **miss** softkey in order to get the Missed Calls list. It indicates the number and date and time. Then navigate through the list with the navigation key to the number to dial, and press **dial** key. Note it is also possible to **delete**, **edit** or **save** the entry.

3.1.2 Call Forwarding

Call Forwarding can be set in three modes:

- Call Forwarding Unconditional: All incoming calls will be forwarded to the number indicated (e.g. Voicemail)
- Call Forwarding Busy: If user is busy (on another call or DND), incoming call will be forwarded to the number indicated (e.g. independent from other CFWD numbers)
- Call Forwarding No Answer: If user is busy with Call Waiting or is not present and does not answer, incoming call will be forwarded to the number indicated, after an specified amount of time (CFWD No Answer Delay).

[13] Dial plan for external calling is configurable; in this example the external line is reached dialing 9. For dial plan configuration options check the SPA9000 user guide.

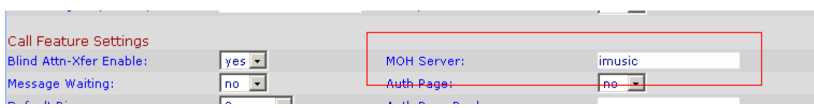
In order to set the call forwarding follow the steps below:

1. Press Menu/setup button ( ) and option 6 – **Call Forward** to enter the Call Forward Menu.
2. Select the Call Forwarding Option: Unconditional (*CFWD AllNumber*), On Busy (*CFWD BusyNumber*) or No Answer (*CFWD No AnsNumber*), and press **edit.** If you would like to modify the CFWD no answer delay (by default 20 s), select *CFWD No AnsDelay*.
3. Enter the Forward-to number (or delay) and press **ok**. By default, voicemail (vm) is selected as the Forward-to number for the On Busy and No answer cases.
4. Press **save**.

In order to test it, call the phone with CFWD settings. Call should be forwarded to the indicated destination (please make sure to configure a valid and reachable destination).

3.1.3 Music on Hold - MOH

Music on Hold is used when parking/holding/transferring a call. Music can be internal music generated by SPA9000 client (named *imusic*) or can be online. This example shows internal imusic. In order to have internal music enabled, make sure **MOH Server** parameter in the phones (Line 1) is set to *imusic.*



In order to demonstrate MOH, perform the following actions (e.g. ALBERTO-103 calls MARIA-101):

1. ALBERTO dials 101
2. MARIA answers and call is established
3. MARIA put ALBERTO On Hold by pressing the Hold key ( )
4. ALBERTO will start hearing the imusic.
5. In order to resume the call, MARIA press the Line key (it should be red and blinking).
6. Call is resumed.

3.1.4 Personal Directory

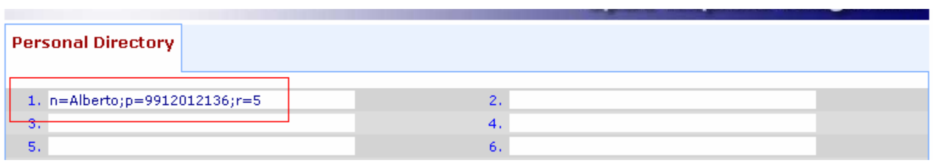
Personal directory includes up to 100 phone book entries. Three parameters can be stored per entry: Name, Number and Ring (used when this user calls you).
There are two ways of storing the numbers:

1. Via phone display
    a. Press **dir** button
    b. Select *New Entry* and press **add** softkey.
    c. Enter Name
    d. Enter Number

e. (optional) Modify ring. In order to modify ring press **option** softkey, choose the ring of your preference (there is the possibility to play it) and press **select**.
f. Press **save**

2. Via Web user interface
   a. In the web user interface, click on the personal directory link, as shown below:

   b. In the personal directory entry, insert the Name (n), Number (p) and ring[14] (r) following the format shown below:



   c. Click on **submit changes** button.

In order to dial a number in the personal directory, press **dir** softkey and navigate through the directory in order to find the contact.

3.1.5 Speed Dial

Speed dial allows setting up to 8 short dial numbers (from 2 to 9) to contacts in the personal directory[15] or numbers entered in the speed dial menu.

In order to set the speed dial code:

1. Press Menu/setup button (🗎) and option 2- **Speed Dial**, then press **select** soft key.
2. Press select the dial number to assign (from 2 to 9) and press **edit**
3. Enter the name of the contact (in the personal directory) or directly the number[16]. The phone will automatically perform discovery against the personal directory, trying to match it to an assigned name, and press **ok** softkey.

In order to make a call using speed dialing, dial the number key (e.g. 2) and press **dial**. An alternative is to enter into the speed dial menu (menu/setup+2), select the speed dial entry and press **dial** softkey.

[14] If you want to use default ring, it is not necessary to incluye the ring (r) parameter.
[15] Currently Corporate directory assignment to speed dial is not supported.
[16] This entry is alphanumeric, please ensure the number is entered correctly.

3.1.6 Call History

Call History includes a log of (re)dialed, answered and missed calls, each with 60 entries maximum. Each entry includes Number (and name if available) and a timestamp indicating date and time. It is possible to dial, edit and save in personal directory any log entry. In order to access the Call History logs:

1. Press Menu/setup button (🗎) and option 3- **Call History**, then press **select**.
2. There are three options available: 1-Redial list, 2-Answered calls and 3- Missed calls. Select the one of your preference and press **select**.
3. Use the navigation key to position into the entry. Press **dial** to call the number indicated in the entry; press **delete** to delete the entry; press **edit** to edit the number[17], press **save** in order to save the number in the personal directory; press **cancel** to return to previous menu options.

3.2 Advanced Business call services

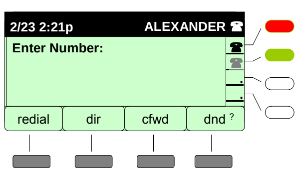
3.2.1 Shared Line

Shared Line operation is as follows. Whenever a user call a shared line (in this example dialing 200 or dialing MAIN number from external to reach and dialing option 0 – Receptionist), all phones sharing the line ring simultaneously.

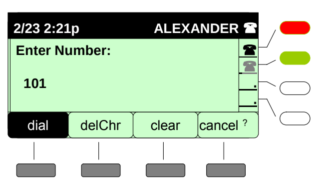
3.2.2 Attended Call Transfer[18] (xfer)

In this example ALEXANDER-102 calls MARIA-101, MARIA-101 calls ALBERTO-103 and indicates she is going to transfer ALEXANDER-102, then executing the transfer.

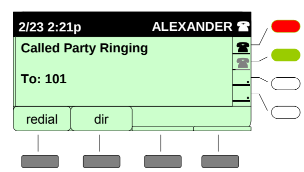
ALEXANDER-102 (A)



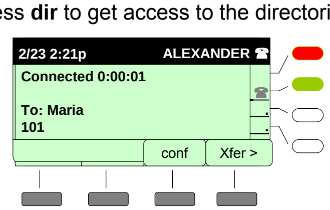
1.- Alexander: (a) Off-hook the phone or (b) press the speaker button, (c) press the line key to get access to the line, or (d) Press **dir** to get access to the directories



2.- Alexander: Enter the number to dial (prefix 9 to reach an external number)
3.- Press "dial" Softkey, or press "#" or wait for the digit timer expiration.

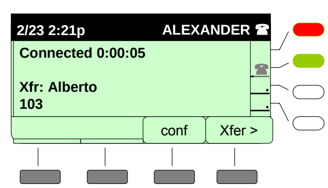


4.- Alexander: Wait for the other end (María-101) to answer



5.- When María Answers, the call get connected

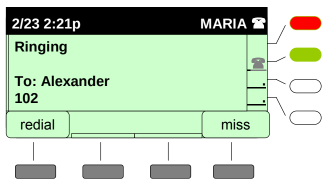
6.- María initiates the attended transfer, so the call get on hold. Alexander get music on hold audio.



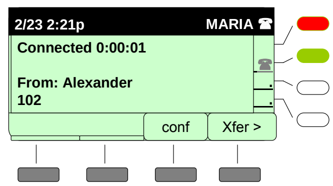
7.-After Maria transfer the call, ALEXANDER get a transfer indication on the display and audio is connected.

[17] Editing the number may be necessary in cases where the number signaled does not match with the dial plan (e.g. in case it comes in international format).
[18] Note this service performs transfer in case both calls are on same line. For transfer using different lines in the phone, another option is available (called **xferLx**) not explained in this demo example.

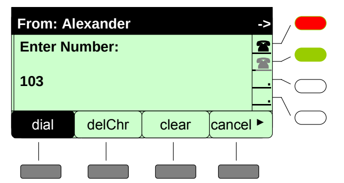
MARIA-101(B)



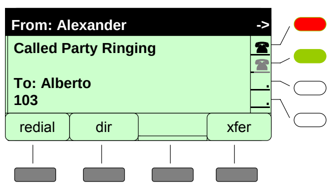
4a.- María: Phone is ringing



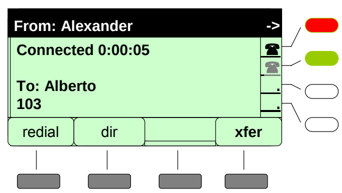
5a.- María answer the phone and call get connected. She decides to transfer the call to Alberto and press **xfer** to initiate the transfer.



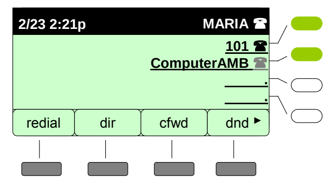
6.- Maria enters the ALBERTO number (103) manually or from **dir**/**redial** menu, and press **dial**



7.- Whilst Alberto's phone is ringing Maria maintains Alexander on Hold

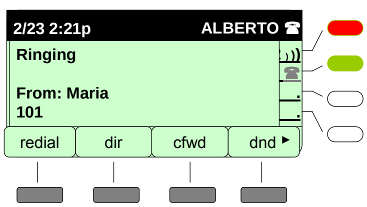


8.- Alberto answers and call is connected. Maria indicates Alberto she is going to transfer Alexander and Maria press **xfer**.

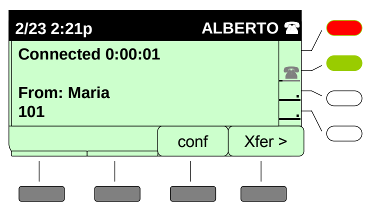


9.- The call is transferred and Maria is now out of the call.

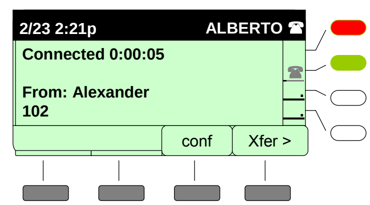
ALBERTO-103 (C)



7a.- After Maria dials Alberto's number (103) it starts ringing



8a.- Once Alberto answers the call is connected. In this moment Alberto knows from Maria she is going to transfer Alexander

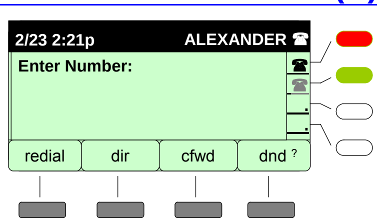


9a.- Once Maria transfer Alexander, Alberto get an indication on the display, indicating the originating party has changed (now it is ALEXANDER)

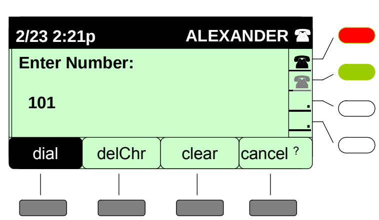
3.2.3 Blind Call Transfer[18] (bxfer)

In this example ALEXANDER-102 calls MARIA-101, which transfers the call directly to ALBERTO-103. ALBERTO-103 will receive the call directly from ALEXANDER-102.

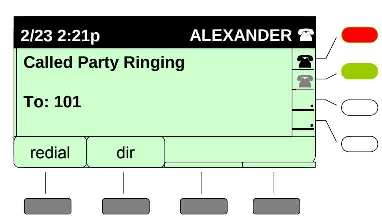
ALEXANDER-102 (A)



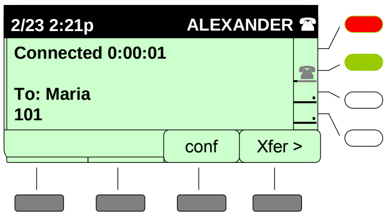
1.- Alexander: (a) Off-hook the phone or (b) press the speaker button, (c) press the line key to get access to the line, or (d) Press **dir** softkey to get access to the personal/corporate directory



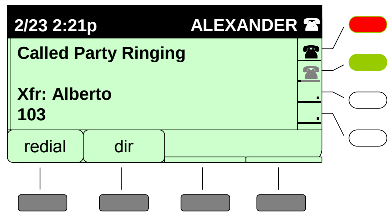
2.- Alexander: Enter the number to dial (prefix 9 if you would like to reach an external number)
3.- Press "dial" Softkey, or press "#" or wait for the digit timer expiration.



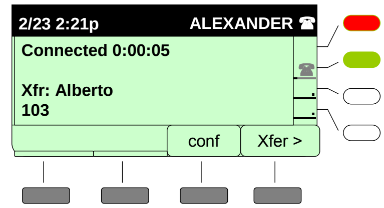
3.- Alexander: Wait for the other end (María-101) to answer



4.- When María Answers, the call get connected

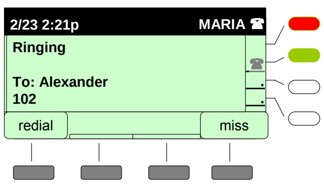


5.- María initiates the blind transfer, and Alexander get automatically the call transferred (ALBERTO is ringing).

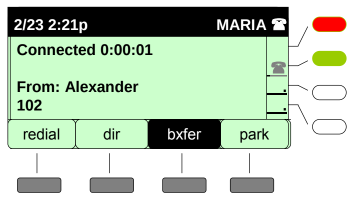


6.-Once ALBERTO answers audio is connected and call is established between ALBERTO AND ALEXANDER.

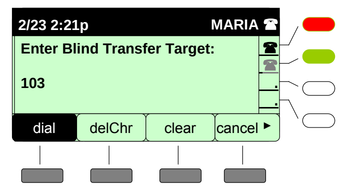
MARIA-101(B)



3a.- María: Phone is ringing

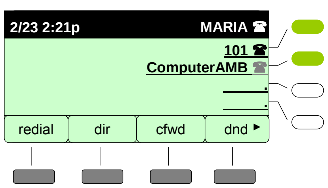


4a.- María answer the phone and call get connected. Maria decides to transfer the call to Alberto, and using the navigation key, press **bxfer** to initiate the blind transfer.

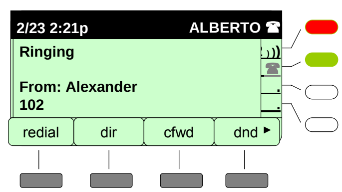


5a.- Maria enters the ALBERTO number (103) manually or from **dir**/**redial** menu, and press **dial**

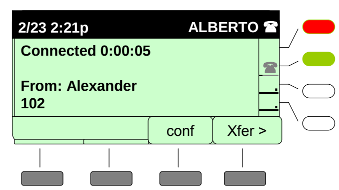
ALBERTO-103 (C)



6a.- The call is transferred and Maria is now out of the call.



5b.- After Maria dials Alberto's number (103) as Blind Transfer destination, Alberto's phone starts ringing already indicating Alexander as originator.

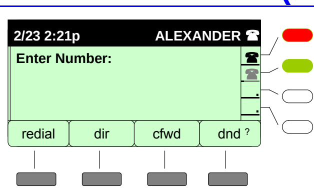


6b.- Once Alberto answers the call is connected.

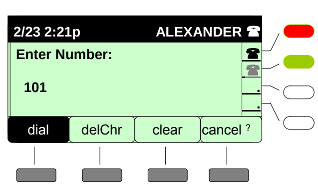
3.2.4 Conference[19]

In this example ALEXANDER-102 calls MARIA-101 and she adds ALBERTO-103 in a conference.

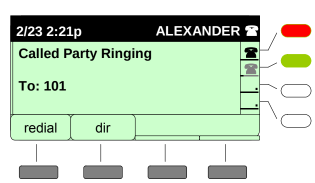
ALEXANDER-102 (A)



1.- Alexander: (a) Off-hook the phone or (b) press the speaker button, (c) press the line key to get access to the line, or (d) Press **dir** to get access to the directory

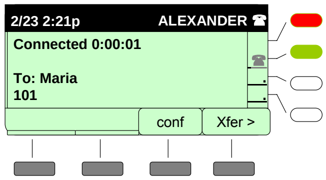


2.- Alexander: Enter the number to dial (prefix 9) to reach an external number)
3.- Press "dial" Softkey, or press "#" or wait for the digit timer expiration.



4.- Alexander: Wait for the other end (María-101) to answer

[19] Note this service performs conference in case both calls are on same line. For conference using different lines in the phone, another option is available (called **confLx**) not explained in this demo example

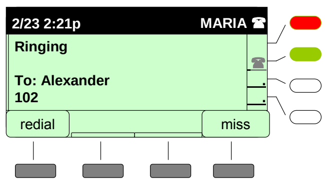


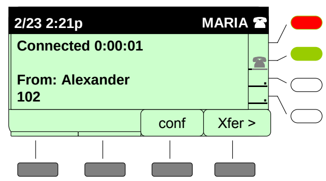
5.- When María Answers, the call get connected

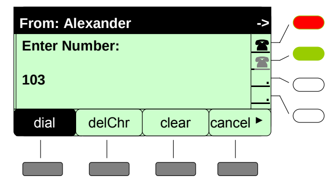
6.- María initiates the conference process, so the call get on hold (whilst Maria is talking to Alberto). Alexander get music on hold audio.

7.-After Maria put call in conference, ALEXANDER get an audible indication and audio is connected between MARIA, ALEXANDER and ALBERTO.

MARIA-101 (B)



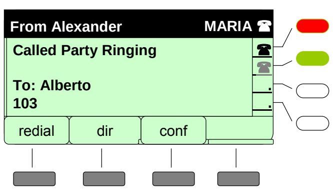


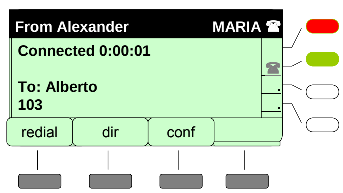


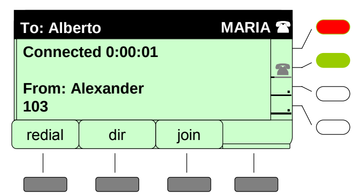
4a.- María: Phone is ringing

5a.- María answer the phone and call get connected. Maria whilst talking decides to conference the call to Alberto and press the **conf** softkey to initiate the conference.

6.- Maria enters the ALBERTO number (103) manually or from **dir**/**redial** menu, and press **dial**





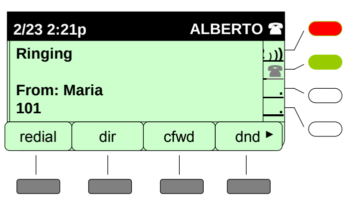


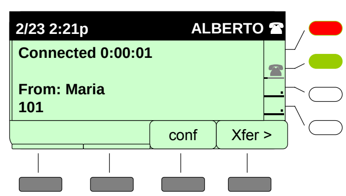
7.- Whilst Alberto's phone is ringing Maria maintains Alexander on Hold

8.- Once Alberto answers the call is connected. In this moment Maria indicates Alberto she is going to conference Alexander and Maria press **conf** softkey.

9.- The call is now in conference.

ALBERTO-103 (C)





7a.- After Maria dials Alberto's number (103) it starts ringing

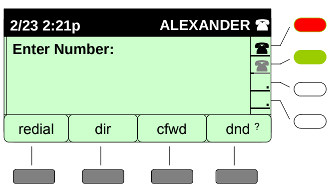
8a.- Once Alberto answers the call is connected. In this moment Alberto knows from Maria she is going to conference Alexander

9a.- Once Maria conference Alexander, Alberto get an audible indication on the phone, indicating the originating party has conferenced.

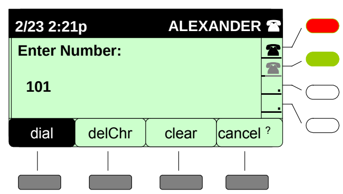
3.2.5 Parking/unparking calls

Parking calls are used to hold a call in one phone and resume the call in a different phone. In this example ALEXANDER-102 calls MARIA-101, she answers the call but would like to talk in a private area, so she park the call, goes to Alberto's desk and unpark it in ALBERTO-103 phone.

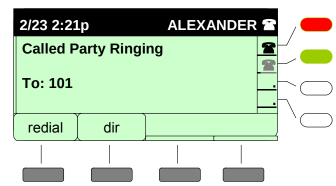
ALEXANDER-102 (Caller)



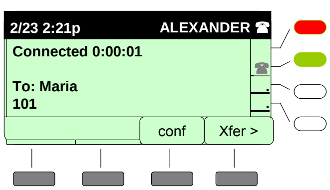
1.- Alexander: (a) Off-hook the phone or (b) press the speaker button, (c) press the line key to get access to the line, or (d) Press **dir** softkey to get access to the directory



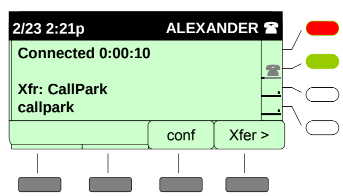
2.- Alexander: Enter the number to dial
3.- Press **dial**.



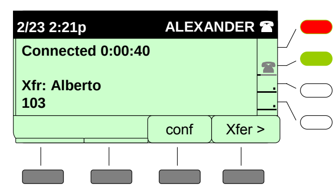
4.- Alexander: Wait for the other end (María-101) to answer



5.- When María Answers, the call get connected

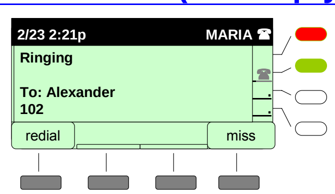


6.- María initiates the call parking, so the call goes to the parking lot (whilst Maria is moving to Alberto's desk). Caller hears MOH.

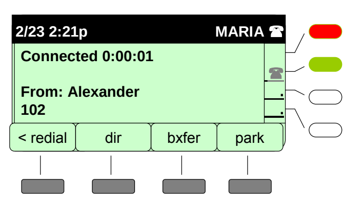


7.-After Maria unpark the call, ALEXANDER is connected to ALBERTO's phone.

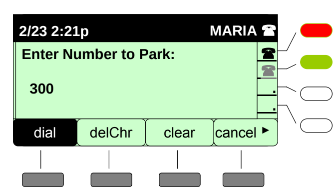
MARIA-101 (Called pty)



4a.- María: Phone is ringing

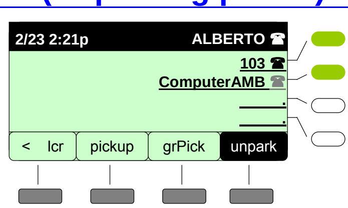


5a.- María answer the phone and call get connected. Maria whilst talking decides to move to other phone and start the call parking process by pressing the **park** softkey.

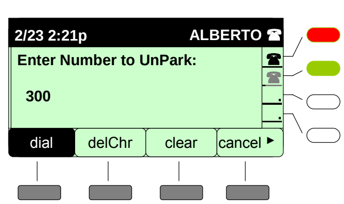


6.- Maria enters the parking lot number (any three-to-four digit number, e.g. 300), and press **dial.** *Call is parked*

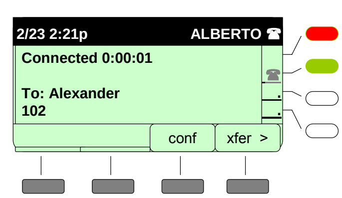
ALBERTO-103 (Unparking phone)



6b.- Maria start unparking call by pressing **unpark** softkey from main menu.



6b.- Maria enters parking lot number (300) and press dial



7a.- Call is connected to ALBERTO's phone.

3.2.6 Call pickup

Call pickup is used in order to pickup a call which is ringing in other phone. If you know which phone (extension) is ringing, use Directed call pickup. If you don't know which phone is ringing, use Group call pickup.

3.2.6.1 Directed call pickup

In this example ALEXANDER-102 calls MARIA-101. Maria is not present, so ALBERTO-103 decides to pickup the call

1. ALEXANDER-102 calls MARIA-101
2. ALBERTO-103 hears MARIA-101 phone ringing and decides to pickup the call, by pressing **pickup** softkey
3. ALBERTO-103 enters MARIA extension number (101) and press **dial** key
4. The phone will display the calls which are actually ringing in Maria's phone (actually one from Alexander)
5. ALBERTO-103 Select Alexander 102 and press **dial**
6. *The call is connected and target phone will stop ringing*

3.2.6.2 Group call pickup

Using similar example (ALEXANDER-102 calls MARIA-101, and ALBERTO-103 hear the ring but cannot identify which phone is ringing):

1. ALEXANDER-102 calls MARIA-101
2. Alberto is listening a ring but cannot identify which station is ringing so decides to do a Group pickup and press **grPick** softkey
3. The phone will show which stations are ringing actually (it will show MARIA)
4. ALBERTO-103 select MARIA and press **dial**
5. The phone then will display the calls which are actually ringing in Maria's phone (actually one from Alexander)
6. ALBERTO-103 Select Alexander 102 and press **dial**
7. *The call is connected and target phone will stop ringing*

3.2.7 Intercom

Intercom lets the user establish a two-way communication directly between two stations. There is no ringing, and audio is opened automatically between the two parties. Any call ongoing in the target is put on hold during the intercom session. In order to perform intercom:

1. Enter *96 and press **dial**
2. The phone will prompt you to enter destination Page target number
3. Enter the extension number and press **dial**
4. Intercom is opened between the stations

3.2.8 Paging

Paging lets the user to establish a one-way communication between the originator and ALL other stations in the system. Audio is opened automatically from origin to destination. Any call ongoing in the target is put on hold during the paging session. In order to perform paging:

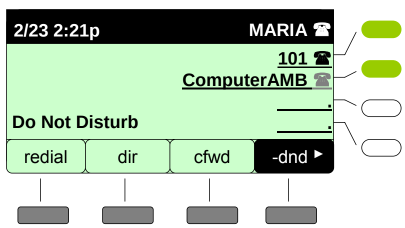
1. Select **dir** (Paging group is an entry in the corporate directory called *PageGroup*).
2. Select *Corporate Directory* and press **select**
3. Position over *PageGroup* and press **dial**
4. Paging session is open

3.2.9 Autoattendant

Playing with the autoattendant is as simple as dialing the MAIN SPA9000 DID (external) number from PSTN. If not available you can dial the autoattendant from any SPA9XX phone, selecting **dir ➔ Corporate Directory ➔ Auto Attendant**

3.2.10 Do not Disturb (dnd)

In order to activate Do not Disturb, press dnd softkey in the main screen. Once activated the sentence Do Not Disturb is shown in the display. When active, any caller trying to reach the user will get Call Ended indication.



Use the **dnd** softkey to activate/deactive Do Not Disturb