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TÍTOL DEL TFC: Implementation and maintenance of telephony network through Avaya s8720: practice application in Fabrics Division at Barcelona.

TITULACIÓ: Enginyeria Tècnica de Telecomunicació, especialitat Sistemes de Telecomunicació

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Overview

W.L. GORE & Associates is an international company based in Barcelona, dedicated to manufacture and sales of PTFE membranes (polytetrafluoroethylene), a breathable and waterproof material with multiple applications on the market. One of these applications is to insert it into bike and running wear, giving extras to garments, improving the performance. Taking this business, we have created an European Customer Service made up to six people with whom we want to offer service to shops and end-users, where they can request and change orders.

Once approved the project in Barcelona, Telecom Department were asked to put it in place, applying the best technology available. To do this, taking benefit of the existing tools in the company, such as our Avasa S8700 telephony server and a customer service of medical products already in service, we will use IP technology to make the logical connections of telephones and PBX with PSTN technology.

The most important part of the project, once ensured a right connection, will be the routing of the calls. This means that we will apply routing sequences to derive the call to the right person to answer the call. In this case, the logic will take into account the origin of the call (European environment) and the person who answers it will know the original language of the caller. To provide some robustness to the system, we increase the complexity of the scenarios where there is no person available to respond the call, the call is received outside office hours, we manage special voice messages, etc.

Having the installation, implementation and system programming completed, we make the corresponding tests for debugging and changes, performing the test period.

Título: Implementación y mantenimiento de la red telefónica mediante el sistema Avaya s8720: aplicación práctica para la división de Fabrics en Barcelona.

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Fecha: 11 de Desembre de 2009

Resumen

La empresa W.L. GORE & Asociados es una empresa multinacional con sede en Barcelona dedicada a la fabricación y venta de membranas de PTFE (Politetrafluoroetileno), un material transpirable e impermeable con múltiples aplicaciones en el mercado. Una de dichas aplicaciones consiste en insertar dicha membrana en prendas de ciclismo y *running*, lo que confiere cualidades extras al tejido y mejora el rendimiento del deportista. Aprovechando esta línea de negocio, se ha creado un departamento internacional de atención al cliente compuesto por seis personas, mediante el cual se pretende dar servicio tanto a tiendas como al cliente final para la solicitud de pedidos y cambios.

Una vez aprobado el proyecto en Barcelona, se encarga al Departamento de Telecomunicaciones de la empresa su puesta en marcha. Para ello, se parte de los medios de los que dispone la empresa, fundamentalmente un servidor de telefonía Avaya S8700 y un *customer service* de productos médicos en activo. En este contexto, se ha utilizado una tecnología IP para las conexiones lógicas de los teléfonos y la centralita de tecnología PSTN. De este modo, utilizaremos telefonía IP en el recorrido entre los teléfonos y la centralita, y pasaremos a modo analógico una vez se haya determinado la gestión de las llamadas, tanto internas como externas.

La parte más importante del proyecto, una vez asegurada la conexión entre los dispositivos, es la programación de los grupos de llamada. En nuestro proyecto, aplicaremos unas determinadas secuencias de encaminamiento para derivar la llamada a la persona más indicada en cada caso. En este escenario, se considerará la procedencia de la llamada (entorno europeo) y la persona que ha de atenderla, que deberá tener un determinado nivel de conocimiento de la lengua de origen de la llamada. Para dotar al sistema de robustez, deberemos determinar un encaminamiento especial o *backup* para aquellos casos en que no haya personas disponibles para atender la llamada, la llamada se realice fuera de horario de oficina, haya que enviar mensajes de voz especiales, etc.

Una vez completada la parte de instalación e implementación, y una vez terminada la programación del sistema, deberán efectuarse las pruebas correspondientes para depurar errores, así como los cambios propios del mantenimiento de las líneas. Igualmente, se han llevado a cabo otras operaciones secundarias como la configuración del sistema en cintas redundantes, la implementación de buzones de voz, la posibilidad de desviar las líneas locales a internacionales, COS y COR y otras opciones adicionales en vectores y skills.

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INTRODUCTION

WL GORE & Asociados is a company which integrates four different divisions: Electronics, Industrial, Fabrics and Medical. Each division works with ePTFE material (Expanded teflon) used for different uses in the market, such as coats, medical products, aerospace, antennas, etc.

This project describes the study and the implementation process which integrates the IP technology in the telephony system of W.L. Gore & Asociados, in the European customer service Fabrics products division. As a result of more than six months of work, we have developed a new customer service in Barcelona office, especially dedicated to respond the needs of the customers for running and bike wear. This department is able to receive calls from all Europe, following internal routing. Basically, the system recognizes where the call is coming from and tries to reach the appropriate person with language skills to answer it (usually, when we call, our expectation is to be understood in our native language).

We use an Avaya switchboard, S8700 server, capable of supporting different types of communication: VoIP, digital and analog. In terms of technology, to upgrade the phones, the company decided to use new IP phones instead of the digital phones currently working, using the traditional PSTN (Public Switched Telephone Network) for internal and external calls. As we are not replacing all the phones (we just add new technology over the current one to provide a new service), we have used Voice over IP (VoIP) from the IP phones to the switchboard, and then we use PSTN to communicate with the exterior and with analog-digital phones. This is a challenge, due to all the technologies have to coexist in the same environment.

The project has been conceived in three stages:

1. IP technology implementation, which includes: IP phones, Switch PoE card (where phones are connected), type of IP media card, VLAN, routing, etc.

2. Configuration of the IP phones and installation, using PoE technology and different features integrated to our phones (customized for our purpose).

3. Programming of the switchboard to work in IP mode, and routing of the lines. This is the most important part of the project, since at this point we really process the call to reach the right person to attend the line. It is based on Avaya software.

The project has consisted in programming a telephony server (or switchboard). Only I participated in the software process and this part haven't got any impact on it. In the hardware, all the materials are the ISO standards without impacting the environment.

CHAPTER 1. IP PHONES

1.1.   IP telephony

IP telephony is a technology which uses Internet protocols and packet-switched connections to exchange voice, fax and other solutions that have traditionally been carried over the dedicated circuit-switched connections of the *Public Switched Telephone Network* (PSTN). Using Internet, calls travel as packets of data on shared lines, avoiding the tolls of the PSTN. The challenge in IP telephony is to deliver the voice, fax, or video packets in a suitable flow to the user. Much of IP telephony focuses on that challenge.
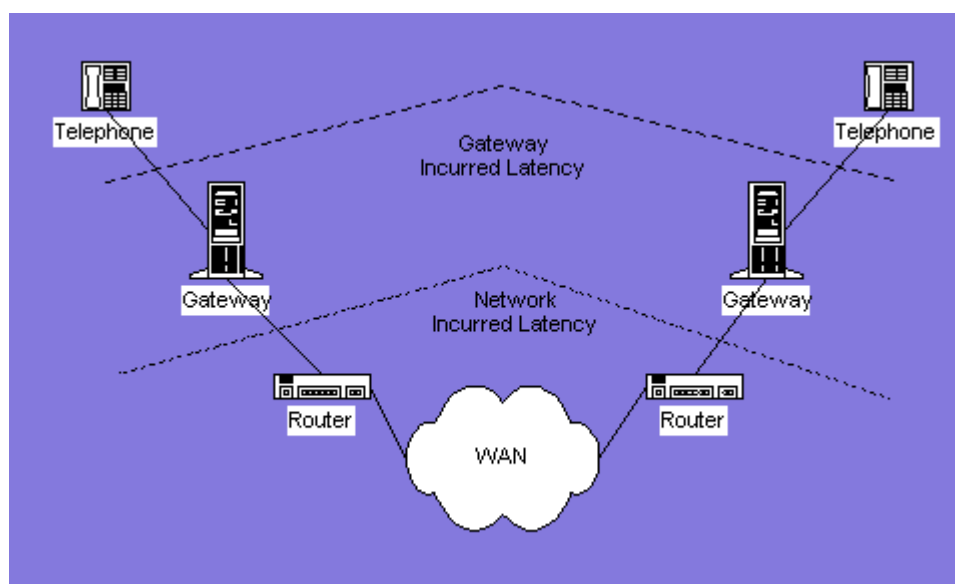
Fig. 1.1 Phone network connection

An IP phone uses *Voice over IP* (VoIP) technologies allowing telephone calls to be made over the internet instead of the ordinary PSTN system. The phones use protocols such as Session Initiation Protocol, Skinny Client Control Protocol or one of various proprietary protocols such as that used by Skype. IP phones can be simple software-based Softphones or purpose-built hardware devices that appear much like an ordinary telephone or cordless phone or an ATA (analog telephony adapter) which allows to reuse ordinary PSTN phones. One of the primary motivations for implementing such a system is the lower calling cost. When calling other IP phones over the internet one only pays for the usually fixed cost internet bandwidth.

Voice over IP (VoIP) refers to transporting voice communication (i.e., "telephone calls") over IP networks, such as the Internet. VoIP is arguably a poor name due to its restricted focus on "voice". The term IP Telephony (IPTel) is more general. Nonetheless, it is the more popular marketing term.

VoIP is viewed as the right choice for provide voice, video, and data communication to people in what is being termed as the Next Generation Network of the future. In fact, all of the circuit-switched equipment vendors and service providers around the world have made plans or are in the processing of planning a migration to VoIP.

It may have many features an analog does not support, such as e-mail-like IDs for contacts that may be easier to remember than names or phone numbers.

1.1.1.   Elements of an IP phone

- DNS (Domain Name System) client. This service translates domain names in IP addresses for traveling into the network. In the final client, a DNS is able to retranslate it in domain server again. The DNS server is a database that contains all the names and the IP address linked.

- STUN client. It is an Internet standards-track suite of methods, including a network protocol, used in NAT (Network Address Translation) traversal for applications of real-time voice, video, messaging and other interactive IP communications. STUN (Simple Transversal of UDP over NATs) is used in some phones to enable the SIP/RTP packets to cross boundaries of two different IP networks. A packet becomes routed between two sip elements if one of the networks uses private IP address range and the other is in public IP address range.

- DHCP client. It may be used to configure the TCP/IP parameters and server details if network segment uses dynamic IP address configuration. DHCP client then provides central and automatic management of IP phones configuration.

- RTP (Real Time Protocol) stack**.** This protocol is providing quality in the packets and it applies feedback to increase the quality of communications like voice, video or both (videoconference). For that, a fast transportation of data is needed without losing quality.

- User interface. It will be the software used by the user for managing the information inside the phone (log calls, directory, options, buttons, etc.)

1.1.2.   Hardware of a stand alone IP phone

Fig. 1.2 IP phone

The overall hardware may look like telephone or mobile phone. An IP phone has the following hardware components.

- Speaker/ear phone and microphone.
- Key pad / touch pad to enter phone number and text.
- Display hardware to feedback user input and show caller-id/messages.
- General purpose processor (GPP) to process application messages.
- A voice engine or a Digital signal processor to process RTP (*Real Time Transport Protocol*) messages. Some Integrated Circuit manufacturers provides GPP and DSP (*Digital Signal Processing*) in single chip. Because for transmitting information on real time, you need a powerful processor of signal.
- Also ADC and DAC converters (to decode it): To convert voice to digital data and vice versa.
- Ethernet or wireless network hardware to send and receive messages on data network.
- Power source, that might be a battery or DC source. Some IP phones receive electricity from Power over Ethernet, using a switch PoE where phones are linked to the switchboard.

1.1.3.   Other devices

There are several WiFi enabled mobile phones and PDAs that come pre-loaded with SIP clients or are at least capable of running IP telephony clients. Some IP phones may also support PSTN phone lines directly.

In analog types, these are usually rectangular boxes that are connected to the internet or Local area network using an Ethernet port and have sockets to connect one or more PSTN phones. Such devices are sent out to customers who sign up with various commercial VoIP providers allowing them to continue using their existing PSTN based telephones.

Another type of gateway device acts as a simple GSM base station and regular mobile phones can connect to this and make VoIP calls. While a license is required to run one of these in most countries these can be useful on ships or remote areas where a low-powered gateway transmitting on unused frequencies is likely to go unnoticed.

1.1.4.   Common features of IP phones

- Caller ID.
- Dialing using name/ID: This is different from dialing from your mobile call register as user need not to save a number to sip phone.
- Locally stored and network-based directories.
- Conference and multiparty call.
- Call park.
- Call transfer and call hold.
- Preserving user name/ number when choosing a different service. provider (not widely supported).
- Applications like weather report, Attendance in school and offices, Live news etc.

1.1.5.   Disadvantages of IP phones

- Requires internet access to make calls outside the local area network unless a compatible local PBX is available to handle calls to and from outside lines.
- IP Phones and the routers they connect through usually depend on mains electricity, unlike PSTN phones which are supplied with power from the telephone exchange.
- IP networks, particularly residential internet connections are easily congested. This can cause poorer voice quality or the call to be dropped completely.

1.2.   Avaya IP 9650 series

This telephone provides many features, including a Phone screen for viewing and managing your calls, 16 auxiliary (Aux) buttons to help you manage calls to bridged lines and additional telephone features, a Call Log, a Contacts List, an integrated WML browser, a menu of options and settings and access to your voicemail.

The Avaya 9650 IP phone is designed to be used in companies dedicated to "Customer Services". Therefore it is used like the main work tool with a lot of functionalities and buttons.

Fig. 1.3 Avaya 9650 IP series used in the project

It supports higher quality wideband audio in both the handset as well as the speakerphone, which provides crystal clear audio with the elimination of background noise. The backlight display and intuitive interface simplifies access to Avaya Communication Manager features. Some features simultaneously manage multiple calls while selectively muting and dropping conference call participants. The 9650 supports built-in button module functionality (eight physical buttons with shift capability for a total of 16 feature keys) to provide simple one-touch access to bridged appearances, speed dials and feature keys. And, with its dual position flip stand, the 9650 makes a smart-looking addition to any desk. Through its integrated web browser and application interface, the 9650 supports productivity enhancing phone applications such as LDAP corporate directories and integration with Microsoft Outlook calendars.

1.2.1. Functions and features

The 9650 features a higher resolution display supporting 1/4 VGA gray scale with backlighting. A four-way navigation button cluster is another feature.

The design facilitates the usage of the display and the built-in browser to improve access to information and use of telephone features. With the enhanced audio across high and low frequencies, it is easier for users to better understand others with different speech.

For the navigator end user, the person constantly on the telephone, the enhanced audio allows calls to be more productive with team members allowing improved collaboration. On conference calls, users find it easier to distinguish and understand multiple speakers, aiding in collaboration and communications. Overall, communications are richer using the Avaya 9650 IP Telephone.

Table 1.1. Description and functions

Name	Description
USB Port	If your phone gets its power from a local AC power adapter (and not directly from the voice/data network connection), you can recharge a battery-powered device using your phone's USB port. Connect the device to your phone with a USB cable to recharge it.
Prompt Line	View the prompt line to see when you can use the right or left navigation arrows to view alternate screens or menus.
Message Waiting Indicator	An illuminated red light in the upper-right corner of your phone indicates you have voicemail messages waiting. If Visual Alerting is enabled, this light flashes when you receive an incoming call.
Line Buttons	The three line buttons with integrated LEDs show which lines are in use and correspond to the lines on the display. Press the line button to select that line. Line button LEDs also indicate if a feature is enabled or disabled in the Feature view.
Softkeys	Use the softkeys to act on objects displayed on the screen. The softkey labels show you the action that each key produces. The labels and the actions vary depending on the object that is selected.
Aux Shift	Use the Aux Shift button to move between one page of eight Aux buttons to the next page of eight Aux buttons. This button lights to indicate a call is active or on hold on the page of Aux buttons not currently displayed.
Aux Buttons	Use the eight auxiliary (Aux) buttons to access up to 16 additional lines, or for advanced telephone features such as Bridged lines, Call Forwarding, Speed Dial, and Abbreviated Dial buttons. Use the Aux Shift button to switch between the first page of eight buttons to the next page of eight buttons and back. The integrated LEDs indicate lines or features in use. White Aux button labels appear in the last two display rows.
Message	Press the Message button to connect directly to your voicemail system.
Navigation Arrows	Use the up and down navigation arrows to scroll through lists. Use the right and left navigation arrows to navigate between menus or to move the cursor during text input.

Name	Description
OK	Press the OK button for a shortcut to an action. For example, when you select a call log entry, pressing OK dials the number.
Phone	Press Phone to view and manage your calls.
Contacts	Press Contacts to view the entries in your contact list.
Call Log	Press Call Log to view a list of your outgoing, incoming, and missed calls.
Missed Call Indicator	The icon on the Call Log button is illuminated when you have missed calls. The top line shows the Missed Call icon and the number of calls missed.
Avaya Menu	Press Avaya Menu to configure options and settings, access the browser, log out, or view network information.
Volume	Press volume to adjust the volume of the handset, headset, speaker, and ringer.
Mute	Press Mute to silence a call in progress. To take a call off mute, press Mute again.
Speaker	Press Speaker to use the speakerphone. To take a call off speakerphone, lift the handset and press Speaker.
Headset	Press Headset to use the headset if it is connected. Only HIS headset cords are compatible with your phone.
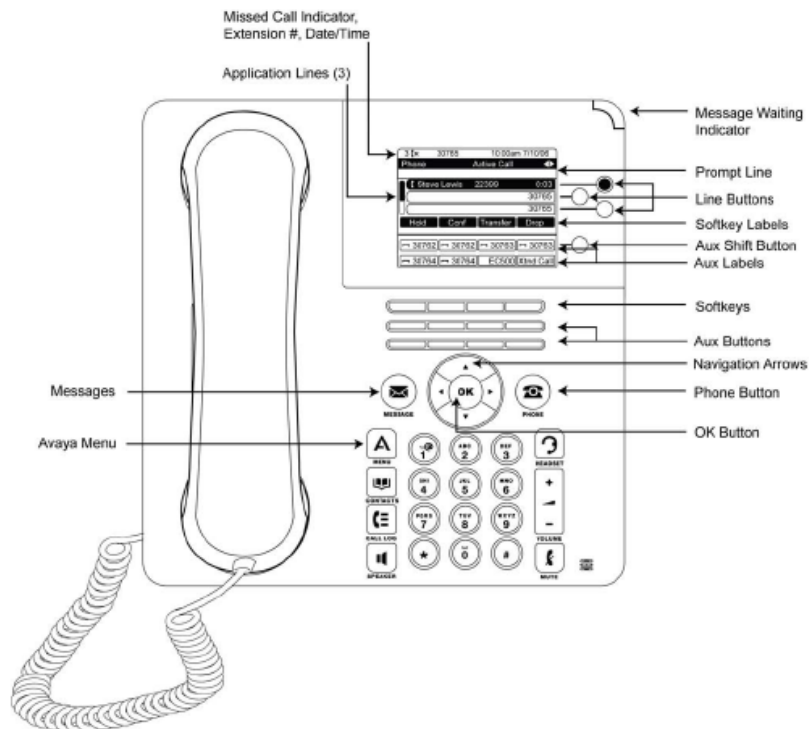
Fig. 1.4 Buttons and functions

1.2.2.   Installing new telephones

We identify three different stages to connect a new phone in the telephony system: find an available port, wire the port to the cross-connect field or termination closet, and tell the telephone system what you're doing.

Before deciding which port you can use for the new telephone, you need to determine what type of device you are installing, what ports are available, and where you want to install the telephone. In our case, we are installing IP phones. As before we were working with analog-digital phones, and it means that we need to change the interconnection technology.

So, we need to add a new switch card for adding IP phones, because IP allows to work using Power over Ethernet. So, the card is able to transfer data and also to supply power for our IP phones. We use the switch PoE summit x250e-24p by Extreme Networks.
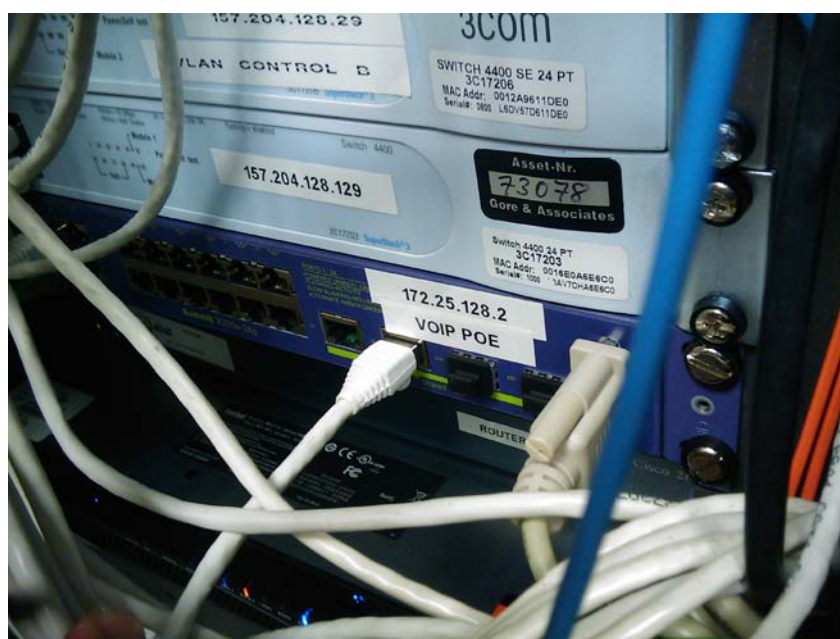
Fig. 1.5 Switch PoE (IP media card)

We have purchased a card with 24 ports (also is available the possibility of 48 ports) where we can connect IP hardware (IP phones Avaya 9650 IP series) and IP softphones (using IP ports).

Once we have connected the Switch PoE to the switchboard, we install our IP phones in the IP media card. The difference between the previous technology is that with the IP phones, we do not need to give a physical port to the station (or phone). When we connect the phone to the switch, we create a new IP station and a virtual port will be assigned directly to the station.

There we go with an example. The first thing to do is to find a hole in our telephony range. We can see the phones already connected with the command *List Station* (later we will see what extensions are occupied by phones, agents, group lines or VDN's).


```
list station                                                    Page   15

                              STATIONS

Ext/       Port/      Name/                     Room/       Cv1/ COR/      Cable/
 Type      Hunt-to    Surv GK NN       Move     Data Ext    Cv2  COS TN    Jack

87042      X          New Agent                              10   1
 603F1                                 no                     1
87043      01A1202    FIT BACKUP                              10   1
 603F1                                 no                     1           D-82
87044      S00028     CHIARA RESSA                        42  10   1
 4620                                  no                     1
87045      S00023     ANNA SUKANY                             10   1
 4620                                  no                     1
87046      S00024     THERESA HAHN                            10   1
 4620                                  no                     1
87047      S00025     DAVIDE BRUZZESSE                        10   1
 4620                                  no                     1
87048      S00026     PABLO RODRIGUEZ                         10   1
 4620                                  no                     1
```

Fig. 1.6 List of stations

It is very important to know the extension we are going to use for our IP phones. We need to know what *Type* of phone model we are going to connect. For adding the new phone, we plug it to the IP media card and we run the Avaya software. Once in, we put the command *Add station XXXXX* (this is the extension number). By default in our Avaya s8700, it is the 6408D+. In the IP phones, we use 9650 series. As this is a newer technology than the switchboard, it does not recognize the model. So, we read the Avaya Administrator Guide to know what kind of phone could be equal to our case.


```
add station 76350                                      Page    1 of    4
                              STATION

Extension: 76350            Lock Messages? n            BCC: 0
     Type: 6408D+            Security Code: _____        TN: 1
     Port: _____       Coverage Path 1: ___          COR: 1
     Name: _____ Coverage Path 2: ___          COS: 1
                            Hunt-to Station: _____

STATION OPTIONS
         Loss Group: 2      Personalized Ringing Pattern: 1
        Data Module? n                 Message Lamp Ext: 76350
       Speakerphone: 2-way           Mute Button Enabled? y
   Display Language: english

                                    Media Complex Ext: _____
                                        IP SoftPhone? n
                                 Remote Office Phone? n
```

Fig. 1.7 Add station

For the part of type of phones, there is a paragraph of the Avaya Administrator Guide of our switchboard where is explained how to recognize our 9650 IP phones. Type is very important (and so when we connect the phone), since depending on the kind of model the functionalities are completely different (buttons, display, configuration, etc.). If we do not write the right model, the system probably will not recognize it.

9650 IP telephone

Communication Manager supports the 9650 IP telephone like the 4620 IP telephone as detailed below, with exceptions as noted:

- Support for the conference, transfer, hold, exit, and drop features
- 24 administrable call appearance/feature buttons per main telephone (16 of the 24 buttons are be displayed on the 9650's auxiliary buttons)
- Support for up to three button expansion modules (SBM24). Each SBM24 allows an additional 24 call appearance/feature buttons per module
- Support for a maximum 96 call appearance/feature buttons between the phone and three SBM24s. (maximum of 10 call appearances or a maximum 96 bridged appearances)
- Speakerphone functionality (none, one-way, and two-way)
- 40 character display message support
- Support for Communication Manager call processing features (except for those features specifically blocked via administration)
- Support of ID Request/Response – with support for a new set ID number of hex code D2
- Support for a new Voice Mail Retrieval function (via the phone Message button)
- Support for update of the Message Waiting Indicator status

Fig. 1.8 Type of station

Now we connect the IP phones and also assign an extension for the different stations.

Table 1.2. Virtual extensions to assign the phones

USER	TERMINAL NUMBER
User 1	87044
User 2	87045
User 3	87046
User 4	87047
User 5	87048
User 6	87049

We use the rank of the virtual number (87XXX) to assign the phone, because we do not use the phone for assigning a line. This is only an instrument of communication, not an important part of our programming because for us it is interesting to assign the lines for the agents (users logged in the system). Later on, we will see the difference between an extension and agent number.

The virtual number only makes sense in internal use and outgoing calls. If we try to call from house to any number 87XXX, we will not be able, because it just follows the logic of our switchboard. Then, we connect an IP phone and now we assign a port. The port is the connection with the switchboard. We have different switches depending on the type of connection that we want to use (IP phones, analog lines, digital lines…). In case of IP phones, we write the type 4620 (equal to 9650 IP series before explained):

Fig. 1.9 Type and port assigned to IP phone

Now the system recognizes the new phone installed in the IP media card. This technology works through IP address and it does not need a fixed way, just a virtual number *S00028* (the "S" at the beginning of the port identifies the device as SIP protocol). Every time that we connect a new IP phone, the port is automatically detected and recognized. It is important to consider our current customer service, the medical division, because it uses Avaya Callmaster IV, that is, analog-digital lines which need fixed ports. See below:

Fig. 1.10 Type and port assigned to digital-analog phone

Here, we must connect in a fix port for the proper recognizing.

Fig. 1.11 Patching

1.2.3.   Phone configuration

Once you have plugged the phones into the network, they start to receive electricity flow by the network cable, due to we know switch PoE send in the same cable power and Ethernet. In the next chapter is explained the PoE technology. So, as we are adding new phones to the switchboard, we have to prepare the scenario for the new phone network diagram.

Fig. 1.12 Phones display

So, we need to assign the IP parameters in network diagram:

Router: 172.25.128.1 (The gate out of our general network in the company)
Switch: 172.25.128.2
Phones: from 172.25.128.6 to 172.25.128.11

We will keep free IP addresses among the switch and IP phones for future expansions in the rack, adding the possibility for new PoE switches with their space in the network.

Required Network Information Before Installation - Per DHCP Server

1. Gateway (router) IP Address(es)	
2. HTTP server IP Address(es)	
3. Subnetwork mask	
4. Avaya Media Server Gatekeeper IP Address(es)	
5. Avaya Media Server Gatekeeper port	Although this can be a value between 0 and 65535, the default value is **1719**. Do not change the default value unless that value conflicts with an existing port assignment.
6. HTTP/HTTPS server file path	
7. Telephone IP Address range *From:* *To:*	
8. DNS server address(es)	If applicable.
9. HTTPS server address(es)	If applicable.

Fig. 1.13 Required table to install IP phones

When we connect these telephones to the network, we give them the IP address and we mark the server to search and to realize the firmware activation. We create a uniform template according to our IP range in Spain. We can see that it has fixed parameters and other variables.The steps to follow are:

Table 1.3. Manual configuration of phones – To assign phones in the network

To access the setting screen on the phone
press:

 MUTE Button   (or , if *"enter code"* is shown just start with 27238 #)
27238 #
ADDR   OK

Change the particular settings for:
(Change process =   *change* > [type value] > save   )

Phone: use a local free IP Adress
CallServer   157.204.162.150   (Extension 1001   pw= 1111 on our G350 test
System)   or use your own CallServer IP 157.204.128.7   and a IP extension
Router: your local default gateway IP address – 157.204.128.1
Mask: 255.255.255.0
File Server   157.204.162.55 to get the current firmware

back > exit   **(do not interrupt download pocess !!! this will destroy phone)**

1.2.4.   Configuring buttons

Now, it is time for programming the buttons. We personalize the phones for the use that they have. As Avaya uses its own OS, there are specific names for the different buttons (for further information, we attach in the annex a table with the different buttons recognized by the Avaya system). In our case, we have:

Fig. 1.14 Button assignments - Lines

For each phone, there are three lines. So, we can handle up to 3 different calls at the same time.

```
 4: abrv-dial  List: 2 DC: 05
 5: abrv-dial  List: 2 DC: 06
 6: aux-work    RC:    Grp:
 7: auto-in           Grp:
 8: after-call        Grp:

 9: q-calls    Grp: 121
10: q-calls    Grp: 122
11: q-calls    Grp: 123
```

Login	Logout	AuxWork	Auto In
AfterCall	Q 121	Q 122	Q 123
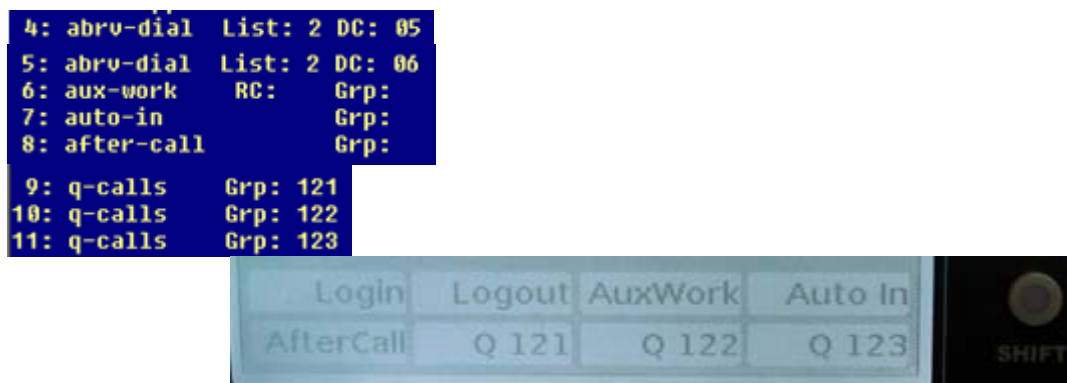
Fig. 1.15 Customized buttons I

The buttons *Login* and *Logout* allow the user to manage the access to the switchboard. It means that the user manually access to the system. Buttons *AuxWork*, *AutoIn* and *AfterCall* show the status of the user into the system.

```
12: q-calls    Grp: 124
13: q-calls    Grp: 125
14: q-calls    Grp: 126
15: q-calls    Grp: 127
16: aut-msg-wt Ext: 76321
17: aut-msg-wt Ext: 76322
18: aut-msg-wt Ext: 76323
19: aut-msg-wt Ext: 76324
20: aut-msg-wt Ext: 76325
21: aut-msg-wt Ext: 76326
22: aut-msg-wt Ext: 76327
```

Q 124	Q 125	Q 126	Q 127
A 76321	A 76322	A 76323	A 76324
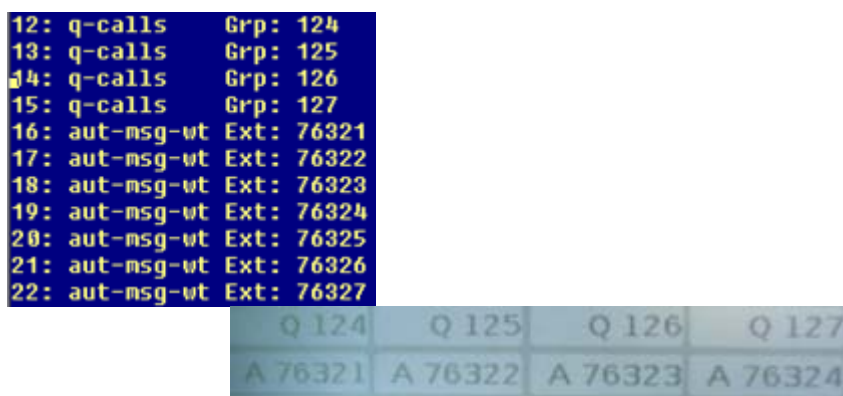
Fig. 1.16 Customized buttons II

Queue Call and Mailbox buttons light the phone when one of them is in use. For the first one, it shows that there is a call waiting in one queue, and for the second one it means there is a message in the mailbox (we can also see the line). Status are:

Aux-work: the user is logged into the system but he does not receive calls (some companies want to know when the user started the day and this is a way to mark the timing). It is also a way to allow the internal communication within the company, using the extension number.

Auto-in: the user is logged into the system and he can start to receive calls from the call group.

After-call: Once finished the call, pushing this button the system goes automatically to Aux-work for 5 minutes and after that, it comes back to Auto-in,

allowing the incoming calls to the agent number again. This is going to enable the person a time to prepare the order of the last call.

1.3.   Switchboard connection

1.3.1.   Power over Ethernet (PoE)

Power over Ethernet (PoE) is a technology for wired Ethernet LANs (local area networks) that allows the electrical current, necessary for the operation of each device, to be carried by the data cables rather than by power cords. This minimizes the number of wires that must be strung in order to install the network. The result is lower cost, less downtime, easier maintenance, and greater installation flexibility than with traditional wiring.

For PoE to work, the electrical current must go into the data cable at the power-supply end, and come out at the device end, in such a way that the current is kept separate from the data signal so that neither interferes with the other. The current enters the cable by means of a component called an injector. If the device at the other end of the cable is PoE compatible, then that device will function properly without modification. If the device is not PoE compatible, then a component called a picker or tap must be installed to remove the current from the cable. This "picked-off" current is routed to the power jack.

To minimize the possibility of damage to equipment in the event of a malfunction, the more sophisticated PoE systems employ fault protection. This feature shuts off the power supply if excessive current or a short circuit is detected.

PoE technology is regulated in a standard called IEEE 802.3af and is designed in a way that does not degrade the network data communication performance or decrease the network reach. The power delivered over the LAN infrastructure is automatically activated when a compatible terminal is identified, and blocked to legacy devices that are not compatible. This feature allows users to freely and safely mix legacy and PoE-compatible devices, on their network.

In our case, as we explained before, we use the switch PoE *summit x250e-24p*, made by Extreme Networks, due to we have currently 6 IP phones to install and also we will connect 6 IP softphones, which they will use IP ports, but in that case, the connection is internally in the switch (it is a logical one) and we will have 12 free ports for future expansions in our telephony system or also if we think to deploy completely IP technology for our office in Barcelona.

Once we have purchased the switch, we need to integrate it in our telephony structure, thinking in two different parts: the network diagram and the physical connection.

1.3.2.   Including switch PoE in the network diagram

Now we study the new network diagram and the IP availables. First, we ask the European network contact for adding this switch, and the need of some IP free to use. After some days, they gave us the possibility to use the rank from 172.25.128.2 to 172.25.128.255 (remember that 172.25.128.1 is marked as the router by default and 172.25.128.255 is used as broadcast), which is already in our private rank of the company. This is the designed network diagram, where we focus in the new component:
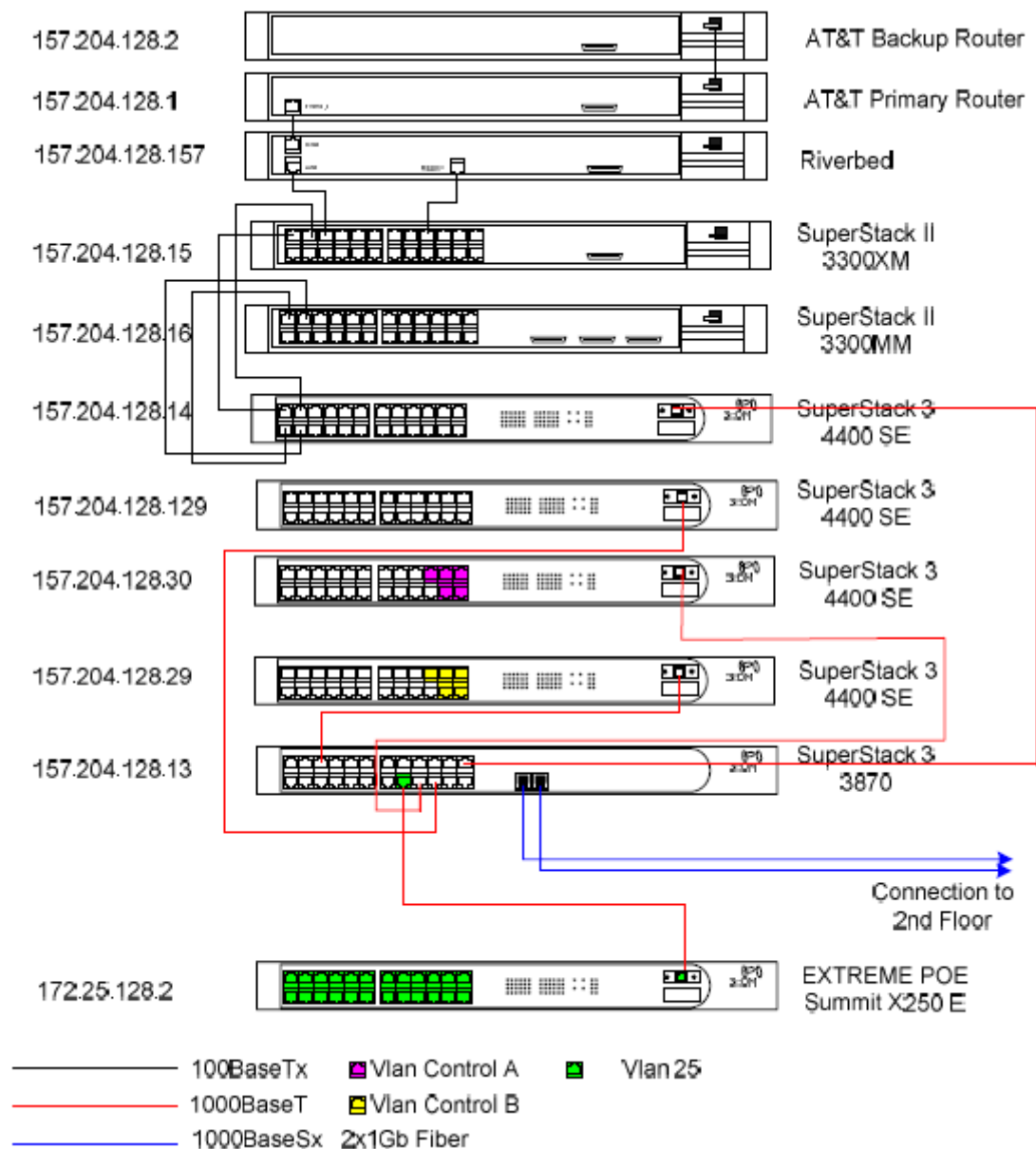
Fig. 1.17 WL GORE – Network Diagram 3rd floor

Now, we connect our IP phones to the PoE, with the IP 172.25.128.2. As we have our Call server and IP media in the rank 157.204.128.XX, we put as router the switch where we can see connected the IP media and the call server, located in the switch 157.204.128.13. Being this scenario now:

Table 1.3. Manual IP configurations

IP	EQUIPO	TARJETA
172.25.128.1	ENLACE SWITCH 157.204.128.13	ETHERNET
172.25.128.2	SWITCH EXTREME POE - 3rd Floor	ETHERNET
172.25.128.3	FREE - FOR FUTURE EXPANSION	ETHERNET
172.25.128.4	FREE - FOR FUTURE EXPANSION	ETHERNET
172.25.128.5	FREE - FOR FUTURE EXPANSION	ETHERNET
172.25.128.6	IP TELEPHONE 9650 STATION 87044	ETHERNET
172.25.128.7	IP TELEPHONE 9650 STATION 87045	ETHERNET
172.25.128.8	IP TELEPHONE 9650 STATION 87046	ETHERNET
172.25.128.9	IP TELEPHONE 9650 STATION 87047	ETHERNET
172.25.128.10	IP TELEPHONE 9650 STATION 87048	ETHERNET
172.25.128.11	IP TELEPHONE 9650 STATION 87049	ETHERNET

As we have ready our diagram, we can start the installation of the switch in our diagram. We put our new switch visible for the switchboard, connecting it to the same IP rank. We explain in next chapter the logic for the Avaya server, when we have different boards prepared to plug IP phones.

1.3.3. Switch PoE: Installation and connection of the hardware

After having the diagram planning in a draft and the IP to use, we can start to install the new switch.
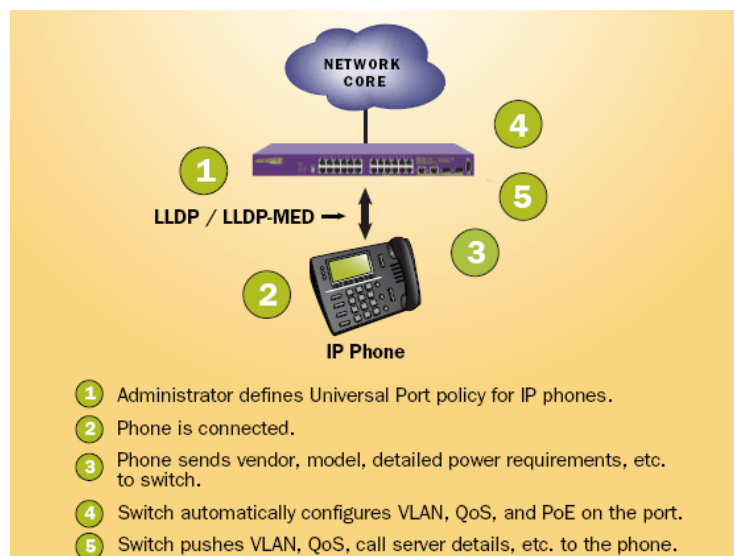
Fig. 1.18 Understanding the connections

CHAPTER 2. AVAYA SERVER S8720

2.1.   PBX

A Private Branch eXchange is a telephone exchange that attends a particular business or office, as opposed to one that a common carrier or telephone company operates for any business or for the general public. PBXs make connections among the internal telephones of a private organization —usually a business— and also connect them to the public switched telephone network (PSTN) via trunk lines. Since they incorporate telephones, fax machines, modems, and more, the general term "extension" is used to refer to any end point on the branch.

Fig. 2.1 Picture of Avaya s8700 server in WL GORE

Inside the boxes we see in Figure 2.2, there are different boards and the controller of the switchboard. We consider the IP media and also the C-LAN, which control the working and running of all the IP stuff.

In WLGORE we have installed three different C-LAN, which are the default gates for the different items connected to it. We want to differentiate within the different boards of use which they require an IP address. In that case, we see:

- CMS supervisor, a program to analize the internal traffic of calls.

- Audix, the mailbox of the different telephones, where we can store the messages when the user is not reachable.

- IP phones, which are all the phones with IP needs.

Fig. 2.2 Boards and IP Media

In practice, when we try to assign manually in the phones the call server, understood like the gate to the switchboard, phones try to get anyone of the mentioned below, because this is just a name and the program will assign randomly the call server.

After that, the important thing is to connect the IP media to our structure and also to be seen by the IP phones. The available IP media is the TN2302, which allows simultaneously 64 connections (good enough since we only will connect 6 different phones). This is the configuration of the IP media and C-LAN's:

Table 2.1. CLAN and IP Media

IP	EQUIPO
157.204.128.5	CLAN 1
157.204.128.6	CLAN 2
157.204.128.7	CLAN 3
157.204.128.8	IP MEDIA CARD
172.25.128.1	ENLACE SWITCH 157.204.128.13

In the table we see as our router for the IP phones is a connection in the switch where we have connected the elements of the switchboard, being totally visible for the phones and vice versa.

In the example, we can see the different switches integrated in the rack. The different connections between them are drawn with a different colour, to see quickly where they are with a simple look.

Fig. 2.3 Boards and IP Media

Fig. 2.4 Patching

Once we have made connections, we test the good running and check if the manual configuration has been applied. So, what we access to Avaya software, and at this point, we put some instructions to see the proper way. We open Avaya Site Administrator and we access to the emulator, we write the instruction *List IP-interface all*, which allows us to see the different elements and the assigned IP's:

```
list ip-interface all
                                 IP INTERFACES
                                                                      Net
ON Type    Slot   Code  Sfx Node Name/      Subnet Mask    Gateway Address  Rgn VLAN
                            IP-Address
-- ----    ----   ----  --- -----------     -------------  ---------------  --- ----
y C-LAN    01A06  TN799  D  clan-cms-only    255.255.255.0  157.204.128.1    1   n
                            157.204.128.5
y C-LAN    01A07  TN799  D  clan-intuity     255.255.255.0  157.204.128.1    1   n
                            157.204.128.6
y C-LAN    01B09  TN799  D  CLN_IP_PBX_1B09  255.255.255.0  157.204.128.1    1   n
                            157.204.128.7
y MEDPRO   01B08  TN2302     IPM01B08        255.255.255.0  157.204.128.1    1   n
                            157.204.128.8
y VAL      01B10  TN2501     VAL01B10        255.255.255.0  157.204.128.1        n
                            157.204.128.9
y MEDPRO   01A09  TN2302     IPM01A09        255.255.255.0  172.25.128.1     3   n
                            172.25.128.4
```

Fig. 2.5 IP interface all

It shows how the IP media is recognized properly by the switchboard, making sense because it was already there. Now, we check if the phones are properly connected, with the instruction **Status station**.

```
status station 87044                                           Page    1 of    5
                              GENERAL STATUS
        Administered Type: 4620            Service State: in-service/on-hook
          Connected Type: 9650         TCP Signal Status: connected
               Extension: 87044
                    Port: S00028        Parameter Download: complete
```
```
status station 87044                                           Page    3 of    5

                          CALL CONTROL SIGNALING

                          Port: S00028
                 Switch                    IP                      IP
                 Port       Switch-end IP Addr:Port   Set-end IP Addr:Port
     IP Signaling: 01A0617  157.204.128. 5   :1720   172. 25.128.  6:2876
           H.245:
       Node Name:           clan-cms-only
   Network Region:          1                         3
```

Fig. 2.6 Status station

The phone is recognized properly, showing the exact addresses that we assigned from the phone when we plugged it in the network. Once we have all the different telephones connected and running in the switchboard, our next step is to create a DHCP server which automatically gives all the network parameters to the phone. We also customize by default the buttons. This is a high task, since we are working with a Novell OS over Windows XP.

2.1.1.   Use of switchboard

At a very basic level, Avaya Communication Manager is a software which administers our phone server, allowing to perform a call processing, and which makes run the system, managing the different accessories (phones, card, ports, etc.). We use the administration interface to let the system know what hardware we have, where it is located, and what we want the software to do with it.

We can find out which circuit packs are in the system and which ports are available by entering the command *list configuration all*. There are variations on this command where are displayed different types of configuration and information. To view a list of port boards on your system we type **list configuration port-network**. The **System Configuration** screen appears.

```
list configuration port-network                                    Page    3

                        SYSTEM CONFIGURATION

Board                                                   Assigned Ports
Number   Board Type             Code    Vintage    u=unassigned t=tti p=psa

01A15    POWER SUPPLY            655A
01B00    POWER SUPPLY            655A
01B01    IP SERVER INTFC       TN2312BP HW15 FW030 01 02 03 04 05 06 07 08
01B02    CALL CLASSIFIER        TN744E  000004     01 02 03 04 05 06 07 08
01B03    ANNOUNCEMENT           TN750C  000008     01 02 03 04 05 06 07 08
                                                   09 10 11 12 13 14 15 16
                                                   17
01B05    DIGITAL LINE          TN2214CP HW08 FW015 01 02 03 04 05 06 07 08
                                                   09 10 11 12 13 14 15 16
                                                   17 18 19 20 21 22 23 24
01B07    ANALOG LINE           TN2215   000017     01 p  p  p  p  p  p  p
                                                    p  p  p  p  p  p  p  p
01B08    IP MEDIA PROCESSOR    TN2302AP HW18 FW111 01 02 03 04 05 06 07 08
```

Fig. 2.7 System configuration

The **System Configuration** screen shows all the boards on your system that are available for connecting phones, trunks, data modules and other equipment. We can see the board number, board type, circuit-pack type, and status of each board's ports. The **u** entries on this screen indicate unused ports that are available for us to administer. These might also appear as **p** or **t**, depending on the settings of our system.

A switchboard is a device used to connect a group of telephones to another one or to an external connection, within and between telephone exchanges or PBX. In case of international companies as W.L.GORE & Associates, the use of switchboards is the best way to interconnect the different locations of our company in the different countries, because it allows to make calls with a short number (extensions) and cheaper than with a normal call (with all the numbers and lines).

The swithboard has also the function of managing the number phones used for the company. The national operator in each country gives a pack of phone numbers, for example in Barcelona, we have all the numbers:

      from   +34934806900 to +34934806999 and
      from   +34933736300 to +34933736350

and then we are the administrators of these numbers in our switchboard.
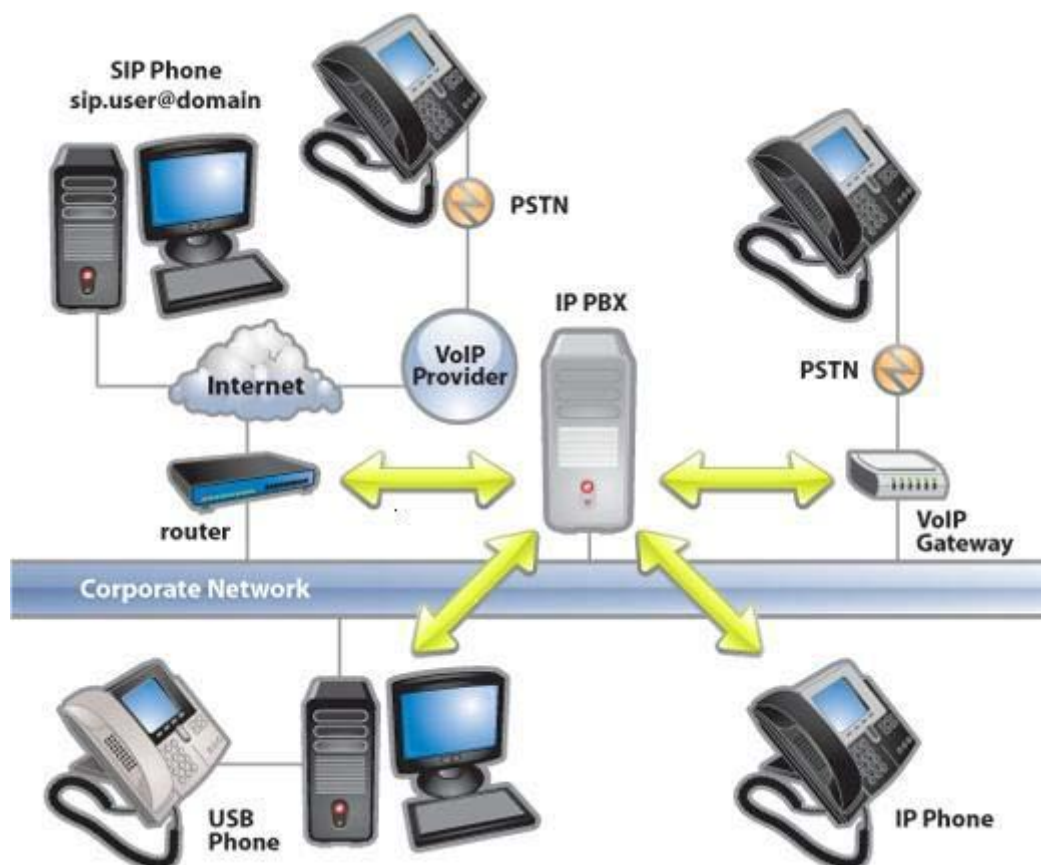

Fig. 2.8 PBX diagram integrating different technologies

This example tries to explain us how a switchboard can interconnect multiple types of terminals and systems (IP and digital phones, internet, PBX's, servers and others).

We give all the details for connecting IP phones to a PBX, and then the managing of outside calls and incoming with the telephony network.

2.1.2.   Basics

First, we need to understand how the users are distributed in the department according to their language skills:

Table 2.2. Organizing people for connections

WINDOW	User 6	User 5	Wear show stand	User 4
	User 1	User 2		User 3

This distribution is directly related to the teams. See the distribution:

User 1: user speaks italian, spanish, english and french.

User 2: user speaks italian, english, french and german.

User 3: user speaks spanish, english and french.

User 4: user speaks italian, spanish, english and german.

User 5: user speaks spanish, english and german.

User 6: user speaks spanish, english and german.

Table 2.3. Routing table

Router	Associate reached	1st backup	2nd backup	3rd backup	4th backup
Germany	User 5	User 2	User 6	User 3	User 4
UK	User 4	User 6	User 1	User 5	
France	User 3	User 2	User 6	User 1	
Italy	User 1	User 4	User 2		
Spain	User 1	User 3	User 4		
Benelux	User 3	User 6	User 1		
Scandinavia	User 4	User 3	User 1		

2.1.3.   System distribution

Next figure shows how gateway manager software can be used to configure and manage gateways that connect the SIP network to other networks, such as the public switched telephone network. This example shows how the gateway manager contains the configuration information for the gateway, including IP address, capabilities such as speech coders, protocols, and access control information.
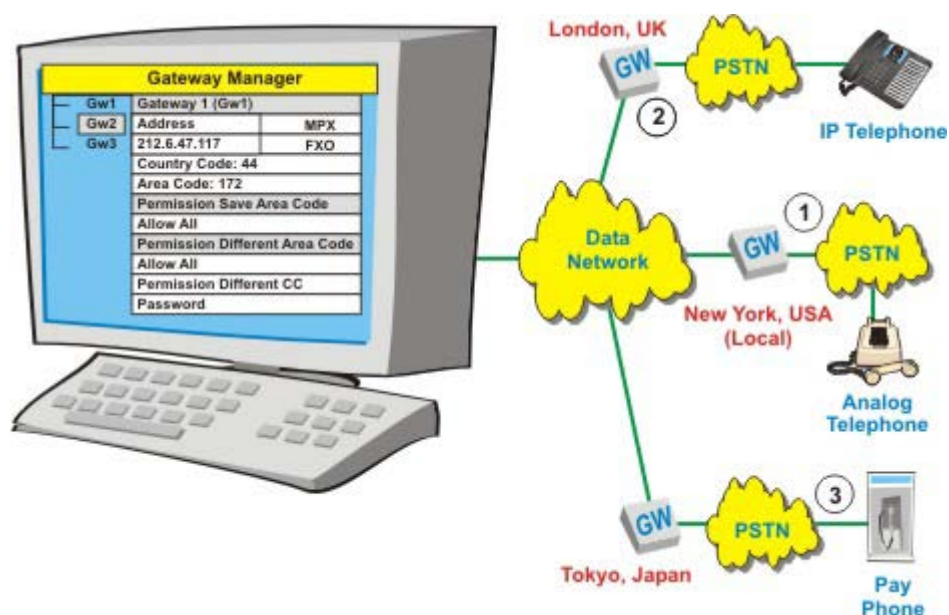
Fig. 2.9 Controlling the PBX

2.1.4.   IP Telephony Communication Servers

Figure 2.10 shows the different types of servers used in a given IP Telephony communication system. This example shows that a call manager (proxy server) receives and processes call requests from communication units (IP telephones). The administrator server coordinates the accounts to the system. A unit manager (location server) functions as a location server by tracking the IP address assigned to the communication units. The gateway manager identifies and coordinates communication through the available gateways. The system manager coordinates the communication between the different servers and programs available on the system.

Fig. 2.10 Telephony diagram – External connection

2.1.5.   The scenario

Preparing the network for IP telephony is essential to a successful migration. Most IP telephony problems, like jitter, latency, and echo, are due to network deficiencies. Here we find some advices about best practices for determining network requirements and testing and evaluating a network prior to a VoIP rollout.

Fig. 2.11 PSTN

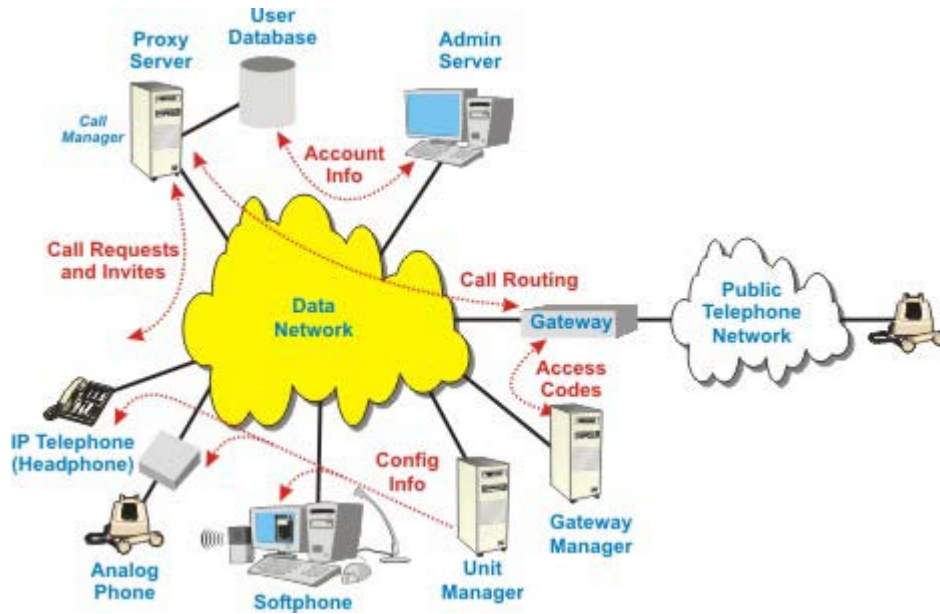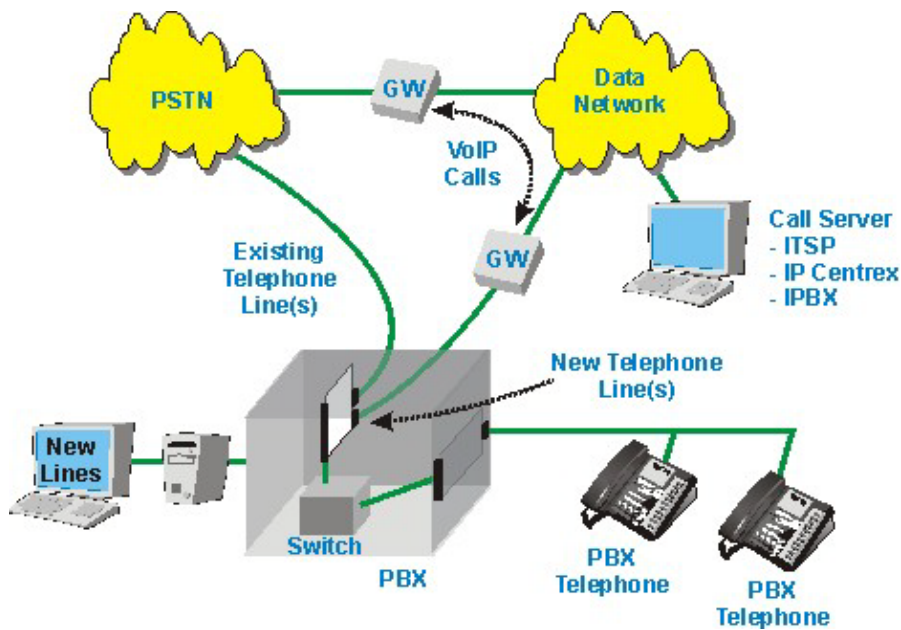Figure 2.11 shows how a PBX system can be upgraded to use voice over IP (VoIP) telephone service without any significant changes to the PBX system. In the example, the VoIP gateway is used to create addition lines for the PBX system. Since the VoIP gateway can produce standard telephone signals (standard telephone or T1/E1 lines), it can be directly connected to PBX system line cards. The PBX system administrator simply configures the PBX system to use the added lines through the same process that would be used when installing new telephone lines. The gateway is controlled to provide telephone services by a call server from an ITSP, IP Centrex or even another iPBX system.

In our case, before we connect the phones, we need to tell the system that we are going to program the new lines and the new stations.

2.1.6.   Stations and extensions

A station is a physical connection that contains a phone line, and so it is a phone for the switchboard. It does exist a big catalogue of models compatible with our Avaya server. In this project, we use the IP phones Avaya 9650. We can create a station using an extension. The extension is a short number that can be understood by the switchboard. So, the VPN is the manager that converts the real numbers (+3493480XXXX) to extensions (76350).

As previous to this project the switchboard was already programmed, we adapt our project to the previous plan. In this case, we program a call center, which means to assign a virtual number at the phones. This kind of system needs to give a number for the station, although in fact, the phone will be used for a user. So, this user will take a number for logging in the system trough the telephone.

In next example, the extension is a short number tantamount to the phone number:

Fig. 2.12 Extension format for agents

They have in common 4 digits at the end. We force to follow a certain logic and to connect properly all the different parts. Anyway, the system does not force to use this planning.

Our agents (users connected to the switchboard) need a terminal number (to activate the phone physically) and the agent number (for logging in our system). We have to understand this as when a user log out to the system, the phone does not disappear, and this needs to keep track to the network. Among other

things, because if the phone is disconnected, it would lose its configuration. Also, in this manner we allow to use the phone to other people, always keeping the same configuration (all the users of this department can use any phone, because as we see later, they have the same profile: buttons, message lights, etc.).

Table 2.4. To assign phone, agent and extension number

USER	TERMINAL NUMBER	PHONE NUMBER	AGENT NUMBER
User 1	87044	934806311	76311
User 2	87045	934806312	76312
User 3	87046	934806313	76313
User 4	87047	934806314	76314
User 5	87048	934806315	76315
User 6	87049	934806316	76316

2.2.   Avaya Communication Manager

2.2.1.   Trunks

A trunk is used to define how telephone signals are carried from one place to another. To understand the meaning of this properly, we itemize with examples the group types, for instance, Integrated Services Digital Network (ISDN). It allows voice, data, video and signalling information to be sent with calls, by connecting telephones, personal computers and other desktop devices to the same system.

Occasionally, your company's telephone requirements change and we need to add new trunk groups to the system. When we need to define a new trunk group, first we must to decide for what it will be used and after we arrange who will be the members of this group (the ports reserved to that performance). Into each trunk, we can choose the multiple options and the service to offer.

In our case, we have a trunk only used for outside calls when we call from the office to a mobile number. In this case, the call is routed into the switchboard unto a SIM card. So, in this case we use a mobile number (6XXXXXXXX) instead a phone number (93XXXXXXX). This is an example of trunk.

2.2.2.   Announcements

An announcement is the recorded message that a caller hears while the call is in a queue. An announcement is often used in conjunction with music. Three types of announcements are:

- Delay announcement: explains the reason for the delay and encourages the caller to wait.

- Forced announcement: explains an emergency or service problem. Used when we anticipate a large number of calls about a specific issue.

- Information announcement: gives the caller instructions on how to proceed, information about the number called, or information that the caller wants.

2.2.3.   Hunt groups

A hunt group is a group of extensions that receive calls according to the call distribution method. When a call is made to a certain telephone number, the system connects the call to an extension in the group. If one person of the group is not available, the call can be answered by other member of the group.

2.2.4.   Vectors

A vector is a set of commands that tells the system how to handle incoming calls. A vector can contain up to 32 steps and allows customized and personalized call routing and treatment. We use call vectoring to:

- play multiple announcements
- route calls to internal and external destinations
- collect and respond to dialed information

The vector follows the commands at each step in order. The vector processes the step and follows the command if the conditions are correct. If the command cannot be followed, the vector skips the step and processes the next step.

The system can handle calls based on a number of conditions, including the number of calls in a queue, how long a call has been waiting, the time of day, day of the week, and changes in call traffic or staffing conditions.

2.2.5.   VDN

A Vector Directory Number (VDN) is an extension that directs an incoming call to a specific vector. This number is a "soft" extension number that is not assigned to a physical location or piece of equipment. VDNs must conform to our dial plan.

CHAPTER 3. PROGRAMMING VOICE LINES

3.1.   Introduction

Once we have all the information about the hardware and how to connect it, we prepare all the new system, programming the different lines.

When we connect the new IP phones to our Avaya telephony server, we configure the IP address in each phone. Each terminal has also a virtual number assigned to the switchboard and each user will have a personal number that is identifyed like agents. This is the table of the first step before programming, planning the new department and giving names to use in the project.

Table 3.1. Phone and IP addressing

USER	TERMINAL NUMBER	PHONE NUMBER	AGENT NUMBER	IP ADDRESS
User 1	87044	934806311	76311	172.25.128.6
User 2	87045	934806312	76312	172.25.128.7
User 3	87046	934806313	76313	172.25.128.8
User 4	87047	934806314	76314	172.25.128.9
User 5	87048	934806315	76315	172.25.128.10
User 6	87049	934806316	76316	172.25.128.11

Then, we have the information to begin the plan in the phones, configuring the IP phones on the network and assigning users with hardware.

After this, we need to configure the call groups. In this case, we separate the different groups by country, and the users can be in various groups. We have in the next table the configuration of the routing calls by teams.

Table 3.2. Making the logic of routing by agent

Router	Associate reached	1st backup	2nd backup	3rd backup	4th backup
Germany	User 5	User 2	User 6	User 3	User 4
UK	User 4	User 6	User 1	User 5	
France	User 3	User 2	User 6	User 1	
Italy	User 1	User 4	User 2		
Spain	User 1	User 3	User 4		
Benelux	User 3	User 6	User 1		
Scandinavia	User 4	User 3	User 1		

So, when a call is routed to German team, the system tries to transfer the associate reached. If it is not available (logout or busy), it routes to the backups.

Now we prepare what will be the new lines to reach. The European Inside sales is doing this part of the preparation because she knows how the system works, so far. With the information taken, we have a phone numbers and the names of the lines by team.

Table 3.3. Preparing the different groups, assigning vector and VDN

Lines	Numbers	Extension or VDN	Vector
FIT General	933746320	76320	20
FIT Germany	933746321	76321	21
FIT UK	933746322	76322	22
FIT France	933746323	76323	23
FIT Italy	933746324	76324	24
FIT Spain	933746325	76325	25
FIT Holland	933746326	76326	26
FIT Scandinavia	933746327	76327	27

All this lines perform the backup planning (explained above). In these lines, we will attend the customers or product consumers with medium priority that wants to propose doubts or attention about products, and the Key Customers with high priority (shop owners). Later, in the examples of vector we find more information about priorities in queues.

We will never call directly to these numbers. Only the system uses this number for routing the incoming calls to the appropriate line (depending on its origin).

Besides, in this department we need additional lines for other kind of callers, the field sales, our associates who are selling textile products. See below:

Table 3.4. Preparing the personal lines

Lines	Numbers	Extension or VDN	Vector
FIT FS User 1	933746328	76328	28
FIT FS User 2	933746329	76329	29
FIT FS User 3	933746330	76330	30
FIT FS User 4	933746331	76331	31
FIT FS User 5	933746332	76332	32
FIT FS User 6	933746333	76333	33

Those lines are addressed directly to an user, because everyone takes field sales by country or regions.

3.2.   Use of Site Administrator: Avaya software

Now we start the assembling of all items previously explained. With a screenshots of the software, we give some details of the process. First, we need to create twelve lines. We show the German line as example and the rest is in the annex.

In the first display of this new software we find all the information that we normally use to configure some parameters of the lines or the switchboard. We have the image below.
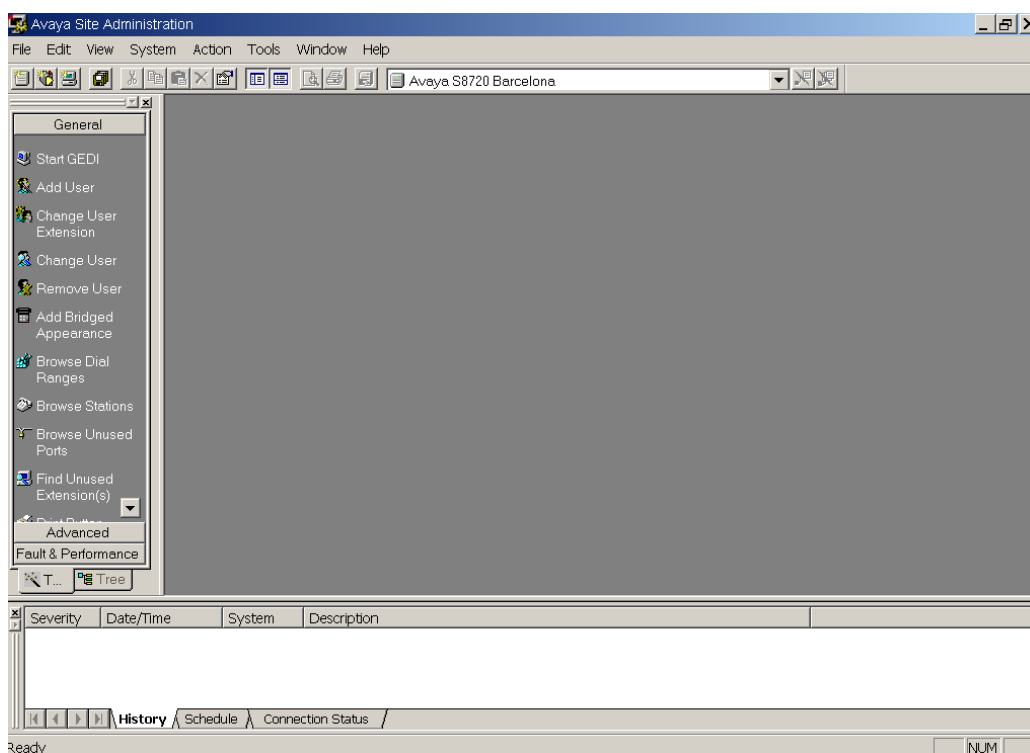
Fig. 3.1 Site administration: Initial view

This is the first window when we open the program:

- On the top, we have the typical information as in any program (File, Edit, Window) and several tabs with specific information (System, Tools). We have also some information in icons.

- On the left part, we have three tabs: General, Advanced and Performance.

▪ General:
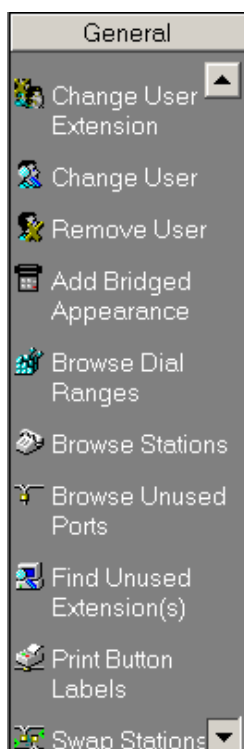
Here we can program the switchboard. We can see the users, the stations and general information about the terminals hardware.

This is a quick start if you do not want to access to general voice lines and skills. This is very useful when we do not have enough knowledge of the system. We can apply basic changes in the users or in the stations (like remove, add or change, browse, swap and others).

Fig. 3.2

▪ Advanced:

In the Advanced options, we can Start Emulation for entering to the general system and program the switchboard. In our case, we use this option to configure lines, stations and agents (users). We can also export data as excel files or text files.

We can modify the templates (for skills, vectors and generate accountings).
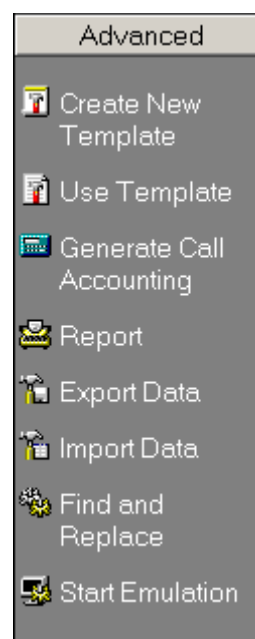
Fig. 3.3

- Performance:
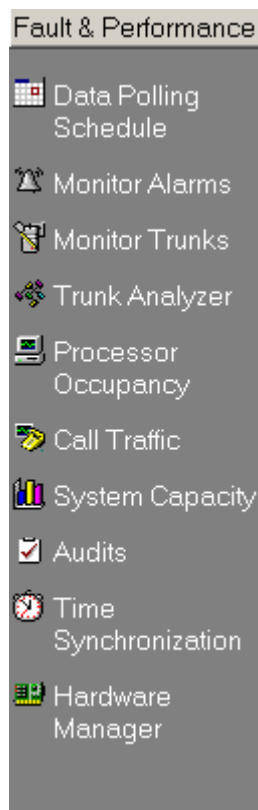
In this tag, we can program a schedule data backup, see the failure logs, analyze the different trunks that form the system structure (incoming and outgoing calls, types of call). Also, we can view the capacity of the system (an interesting option for the reporting).

We can also manage the hardware (Hardware Manager). Normally, when we add a new user or terminal, we need to know where we must connect the terminal (the IP media card in our case) and the free ports in every telephony card.

These systems always include a timing or clock, because the companies have usually a timetable and we should know when the switchboard needs to activate the system. The timing option is inside Time Synchronization.

Fig. 3.4

3.2.1.   Opening the software

We open the Avaya Site Administration and got to pull down Advanced. Here we have the icon Start Emulation and in that moment, the software tries to connect to the system.

Fig. 3.5

Once we enter in the system, with the switchboard IP we see Figure 3.6.
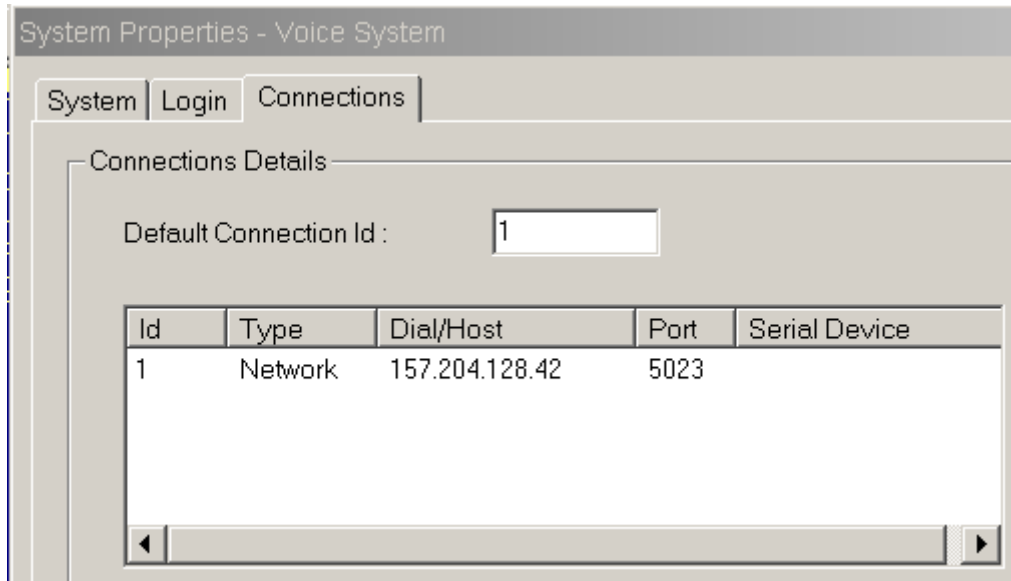
Fig. 3.6

Now we enter on the administration software. In this screen, we can write the commands to administer or to configure any option of the system. In next sections, we explain the most common commands used in the project.
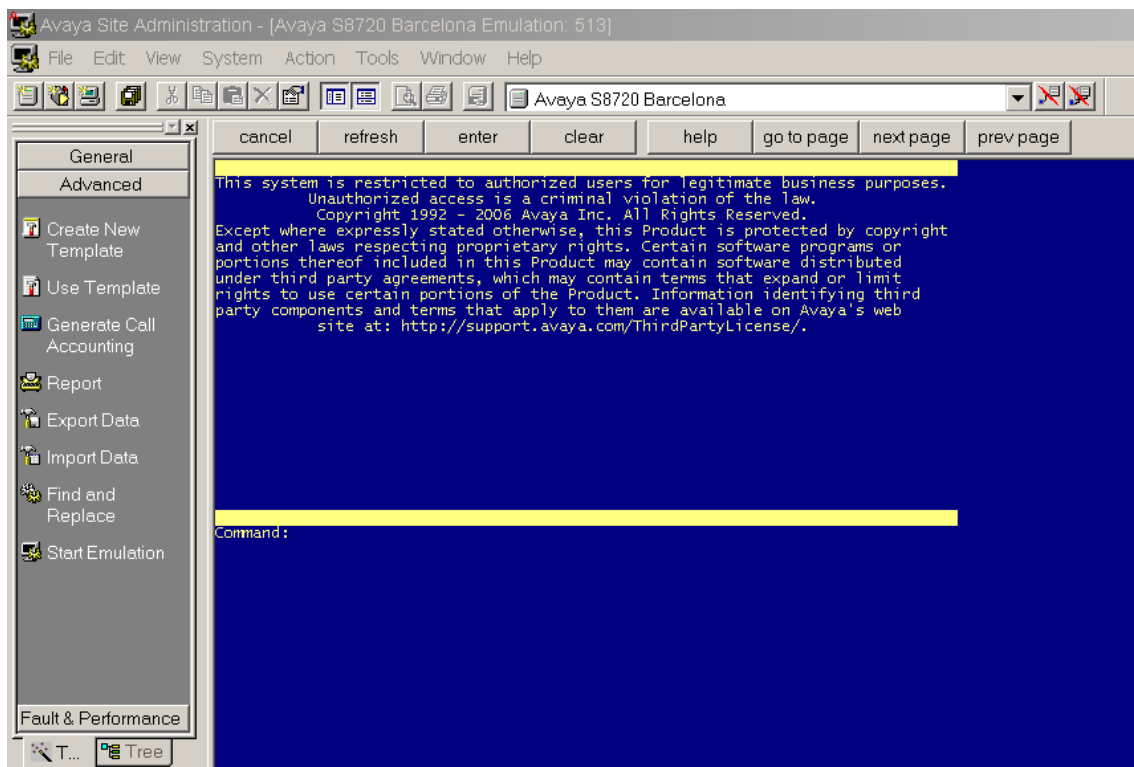
Fig. 3.7

3.3.   Customizing functions

3.3.1.   Vector routing tables

In the first display of this new software we have all the information that we normally use when we need to configure some parameter of the lines or the switchboard. We have the next list:

```
list vrt                                                        Page   1

                        VECTOR ROUTING TABLES

        Number  Name               Sort  Number of Entries

           1    GERMAN INTERVEN     n     100
           2    GERMAN Table 2      n     70
           3    SPAIN               n     0
           4    ITALY               n     0
           5    UK                  n     0
           6    FRANCE              n     0
           7    SWEDEN              n     0
           8    BELGIUM             n     0
           9    NETH DUTCH          n     0
          10    FIT COUNTRY COD     n     20
          11    FIT KA GERMANY      n     61
          12    FIT KA UK           n     41
          13    FIT KA FRANCE       n     5
          14    FIT KA ITALY        n     11
          15    FIT KA SPAIN        n     36

         press CANCEL to quit --  press NEXT PAGE to continue
```

Fig. 3.8

In the first screenshot we can see how is the formal display in Site Administrator (Avaya software for managing the switchboard). From now on we only show the important part to detail.

```
list vrt

                  VECTOR ROUTING TABLES

     Number  Name             Sort  Number of Entries

       16    FIT KA HOLLAND    n     0
       17    FIT KA SCANDINA   n     2
```

Fig. 3.9

VRT's contains different numbers that the system needs to route in a specific way. In our project, we use that for filling in with numbers of Key Accounts divided by countries. When a number of the list is trying to call us, the telephony system recognizes it and routes the call using the vectors. We give country names depending when we want to use it.

The picture shows how many numbers has every VRT. In our example, the routing table of Scandinavia contains only two numbers. We can see the numbers format.

```
display vrt 11                                        Page   1 of   3
                          VECTOR ROUTING TABLE

     Number: 11        Name: FIT KA GERMANY       Sort? n

          1: 00499572912054              17: 004928712755284
          2: 00498966540                 18: 004909114804924
          3: 00498966541275              19: 004906218755390
          4: 00498966541224              20: 0049413788124
          5: 00498966541335              21: 0049421992039
          6: 00498966541216              22: 00499119303524
          7: 00498966541655              23: 0049620112156
          8: 00498966541120              24: 0049303030670
          9: 00498966541131              25: 00494215227724
         10: 00498966541111              26: 0049865448800
         11: 00492871275582              27: 004922258888128
         12: 00492871275571              28: 0049228978480
         13: 00492871275522              29: 00496942603655
         14: 00492871275591              30: 00497115779939
         15: 004928712755450             31: 004962049860492
         16: 004928712755288             32: 004962117990050
```

Fig. 3.10

3.3.2. Hunt example

We use the German line as example. In this case, we need to create a hunt group number and we decide to use the number 121 because the hunt 21 is taken by other lines (lines for Medical Division). The hunt groups for the rest of the countries continuous after this.

When we create a new hunt group, the great part of the parameters is activated by default. In our project, we must change specific parameters dedicated exclusively for customer services divisions.

Hunt groups are composed by three pages. Each page contains different information about the lines. In the first page we have information about the line and the type of calls that it will receive. The parameters that we change are:

- Vector in "y" (yes), because we will use vectors with the code lines.

- Queue in "y". The system allows to place incoming calls in queue when all the available agents are busy. Though the length of the queue is unlimited, the maximum number of calls in our division at the same time can be 8 or 9 for 6 users, which means 2 or 3 calls on the queue.

- Calls Warning Threshold in "1". When a call is put in the queue, the system turn on a light in the telephone of each user. This light button informs then about in what line the call is on hold.

```
display hunt-group 121                              Page   1 of   3
                         HUNT GROUP

        Group Number: 121                           ACD? y
          Group Name: FIT GERMANY                  Queue? y
     Group Extension: 87521                        Vector? y
          Group Type: ead-mia
                  TN: 1
                 COR: 1               MM Early Answer? n
       Security Code:          Local Agent Preference? n
  ISDN/SIP Caller Display:


         Queue Limit: unlimited
Calls Warning Threshold: 1      Port:
 Time Warning Threshold:        Port:
```

Fig. 3.11

We assign virtual numbers (in our case the 87521), because Avaya server recognize numbers, not names and this is the routing reference. This is an important concept since you need to translate properly our information in a language that the software could understand.

The second screenshot shows the quality of service (QoS) and different parameters related to the statistic system (we do not talk in this project about statistics because they use CMS supervisor, another program that contains a higher level of knowledge).

```
display hunt-group 121                              Page   2 of   3
                         HUNT GROUP

              Skill? y      Expected Call Handling Time (sec): 180
                AAS? n         Service Level Target (% in sec): 80 in 10
           Measured: both
  Supervisor Extension:


  Controlling Adjunct: none


     VuStats Objective:
 Timed ACW Interval (sec):
   Multiple Call Handling: on-request



                          Redirect on No Answer (rings):
                                      Redirect to VDN:
            Forced Entry of Stroke Counts or Call Work Codes? n
```

Fig. 3.12

3.3.3.   Example of vector

This is the most important part of the project, because we define the steps to follow when a call arrives to a specific line. In this case, we use as example German FIT line. When a call gets the system, this tries to determine the origin of the call. It recognizes that the call comes from Germany and it routes to the specific trunk. Then it arrives its VDN and this determine the pointer to the vector with what the system does with the call.

```
display vector 21                                              Page    1 of    3
                               CALL VECTOR

    Number: 21                    Name: FIT GERMANY
Multimedia? n        Attendant Vectoring? n       Meet-me Conf? n              Lock? n
     Basic? y    EAS? y    G3V4 Enhanced? y    ANI/II-Digits? y    ASAI Routing? n
 Prompting? y    LAI? y   G3V4 Adv Route? y    CINFO? y    BSR? y    Holidays? y
 Variables? y    3.0 Enhanced? y
01 wait-time     0    secs hearing ringback
02 goto step     24             if staffed-agents   in skill 110         =    1
03 goto step     21             if holiday          in       table 1
04 goto step     16             if time-of-day      is all 18:00 to all 09:00
05 goto step     16             if time-of-day      is sat 09:00 to sun 18:00
06 goto step     26             if ani              in       table 18
07 goto step     29             if ani              in       table 19
08 goto step     32             if ani              in       table 20
09 queue-to      skill 121  pri 1
10 wait-time     10   secs hearing ringback
11 collect       1    digits after announcement 87420     for none
```

Fig. 3.13

The first parameters determine the type of the structure. For example, if the code lines can contain holiday tables (see above the option Holidays with "y"), type of call (multimedia), activation of the meeting conference (not in our case) and other specifications by default.

The next part of this page is divided by lines and in each one there is a step. With this software, it is very easy to understand what every step does. In our case, we will give more details with the most interesting lines and the code used.

- GOTO STEP: we tell the system that if the second part of the command line is confirmed, the call will jump to the line specified.

- TIME OF DAY: with this command we can administer a timetable. Avaya has an internal clock for the timming.

- HOLIDAY: we can determine a holiday timetable. In our situation, we have six different tables for each country.

- WAIT TIME: while the call is on queue, we can offer the system three options: music, silence or ringback.

- QUEUE: we can route the call to a specific skill (or hunt) with a specific priority (low, medium or high). Normally the queues are FIFO (First In First Out) if all the calls takes same priority.

- COLLECT: when we wait to collect numbers from the caller. We use that option if the caller wants to leave a message, because all the available agents are busy or not available, or the caller is calling out of hours.

Let's see the different steps in detail:

```
display vector 21                                               Page   1 of   3
                              CALL VECTOR

     Number: 21                    Name: FIT GERMANY
Multimedia? n      Attendant Vectoring? n     Meet-me Conf? n            Lock? n
     Basic? y    EAS? y   G3V4 Enhanced? y    ANI/II-Digits? y    ASAI Routing? n
 Prompting? y    LAI? y   G3V4 Adv Route? y    CINFO? y    BSR? y   Holidays? y
 Variables? y    3.0 Enhanced? y
01 wait-time     0    secs hearing ringback
02 goto step     24            if staffed-agents   in skill 110        = 1
03 goto step     21            if holiday          in      table 1
04 goto step     16            if time-of-day      is all 18:00 to all 09:00
05 goto step     16            if time-of-day      is sat 09:00 to sun 18:00
06 goto step     26            if ani              in      table 18
07 goto step     29            if ani              in      table 19
08 goto step     32            if ani              in      table 20
09 queue-to      skill 121  pri 1
10 wait-time     10   secs hearing ringback
11 collect       1    digits after announcement 87420    for none
```

```
display vector 21                                               Page   2 of   3
                              CALL VECTOR


12 goto step     19            if digits           =      1
13 wait-time     20   secs hearing music
14 goto step     11            if unconditionally
15 stop
16 collect       1    digits after announcement 87422    for none
17 goto step     19            if digits           =      1
18 stop
19 messaging     skill 42   for extension active
20 stop
21 collect       1    digits after announcement 87421    for none
22 goto step     19            if digits           =      1
```

```
display vector 21                                               Page   3 of   3
                              CALL VECTOR


23 stop
24 route-to      number 900491622645590  with cov n if unconditionally
25 stop
26 queue-to      skill 134  pri m
27 goto step     10            if unconditionally
28 stop
29 queue-to      skill 135  pri m
30 goto step     10            if unconditionally
31 stop
32 goto vector   34   @step 1  if unconditionally
```

Line 1

```
01 wait-time      0    secs hearing ringback
02 goto step     24              if staffed-agents   in skill 110           =   1
```

The system tests if special agent is active.

```
display hunt-group 110                                          Page    1 of    3
                              HUNT GROUP

        Group Number: 110                              ACD? y
          Group Name: FIT SPECIAL AGENT                Queue? y
     Group Extension: 87510                           Vector? y
          Group Type: ead-mia
                  TN: 1
                 COR: 1                      MM Early Answer? n
       Security Code:                   Local Agent Preference? n
  ISDN/SIP Caller Display:


         Queue Limit: unlimited
Calls Warning Threshold:         Port:
 Time Warning Threshold:         Port:
```

Fig. 3.14

```
24 route-to      number 900491622645590  with cov n if unconditionally
25 stop
```

```
03 goto step     21              if holiday          in      table 1
```

Table 1 of holidays is a table where we can add days off to make sure that the system recognizes when is holiday.

```
list holiday-table tablee                                       Page    1

                           HOLIDAY TABLES

              Table Number        Name

                   01             GERMAN HOLIDAYS
                   02             SPAIN HOLIDAYS
                   03             ITALY HOLIDAYS
                   04             UK HOLIDAYS
                   05             FRANCE HOLIDAYS
                   06             SWEDEN HOLIDAYS
                   07             BELGIUM/LUX HOLIDAYS
                   08             NETHERLANDS HOLIDAYS
                   09             CATALONIAN HOLIDAYS
                   10             EXPORT HOLIDAYS
                   11             24 MARCH MOBILE MESSAGE SPA
                   12             FIT SPECIAL HOLIDAYS
                   13             EXPORT ROUTE TO MOBILE
                   14             SCANDINAVIA SPECIAL
                   15
```

Fig. 3.15

```
display holiday-table 1
                              HOLIDAY TABLE

      Number: 1              Name: GERMAN HOLIDAYS

          _____START_____        _____END_____
      Month Day Hour Min       Month Day Hour Min    Description
       01    01   00   00        01    01   23   59   1 JANUARY 2008
       03    21   00   00        03    21   23   59   21 MARCH 2008
       03    24   00   00        03    24   23   59   24 MARCH 2008
       05    01   00   00        05    01   23   59   1 MAY 2008
       05    12   00   00        05    12   23   59   12 MAY 2008
       10    03   00   00        10    03   23   59   3 OCTOBER 2008
       12    25   00   00        12    25   23   59   25 DECEMBER 2008
       12    26   00   00        12    26   23   59   26 DECEMBER 2008
```

Fig. 3.16

The system goes to line 21:

```
21 collect        1    digits after announcement 87421      for none
22 goto step      19             if digits          =       1
```

```
list announcements                                          Page    5

                       ANNOUNCEMENTS/AUDIO SOURCES
Annc.     Announcement                                       Source
Number    Extension    Type         Name                     Combinations
65        87357        integrated   EXPORTHOLIDAYS           1
66        87358        integrated   EXPORTAFTERHOURS         1
67        87359        integrated   EXPORTSPECIALCIRCUMSTANCES 1
68        87410        integrated   EUITACAVAILABLE          1
69        87411        integrated   EUITACNOTAVAILABLE       1
70        87361        integrated   SPAINMOBILECLARA         1
71        87362        integrated   FRANCEMOBILEDIDIER       1
72        87363        integrated   BELGIUMFLEMISHMOBILEBART 1
73        87364        integrated   BELGIUMFRENCHMOBILEBART  1
74        87366        integrated   SCANDINAVIANILLNESS      1
75        87420        integrated   FITGERMANY1STQUEUE       1
76        87421        integrated   FITGERMANYHOLIDAYS       1
77        87422        integrated   FITGERMANYAFTERHOURS     1
78        87423        integrated   FITGERMANYSPECIALCIRCUMSTAN 1
79        87424        integrated   FITUK1STQUEUE            1
80        87425        integrated   FITUKHOLIDAYS            1
```

Fig. 3.17

Then the caller listens a message (recorded by the user which says: today is holiday), we wait him to push a digit and then we collect the message in our mailbox.

Then:

```
04 goto step     16              if time-of-day     is all 18:00 to all 09:00
05 goto step     16              if time-of-day     is sat 09:00 to sun 18:00
```

If the clock system is between these times, it goes to line 16 (that time is when the office is closed, so we create a mailbox).

```
16 collect       1    digits after announcement 87422    for none
17 goto step     19              if digits          =      1
18 stop
```

The announcement 87422 is:

```
77        87422      integrated    FITGERMANYAFTERHOURS              1
```

After the announcement, we go to line 19, when the caller can leave a message.

```
19 messaging    skill 42    for extension active
20 stop
```

```
display hunt-group 42                                    Page   1 of  60
                              HUNT GROUP

            Group Number: 42                      ACD? n
            Group Name: intuity                   Queue? y
         Group Extension: 76998                   Vector? n
            Group Type: ucd-mia          Coverage Path:
                    TN: 1      Night Service Destination:
                   COR: 1              MM Early Answer? n
           Security Code:          Local Agent Preference? n
    ISDN/SIP Caller Display:

             Queue Limit: unlimited
    Calls Warning Threshold:      Port:
     Time Warning Threshold:      Port:
```

```
display hunt-group 42                                    Page   3 of  60
                              HUNT GROUP
         Group Number: 42      Group Extension: 76998    Group Type: ucd-mia
    Member Range Allowed: 1 - 1500    Administered Members (min/max): 1   /6
                                      Total Administered Members: 6
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)        Ext      Name (24 characters)
   1: 76941    AUDIX 1                    14:
   2: 76942    AUDIX 2                    15:
   3: 76943    AUDIX 3                    16:
   4: 76944    AUDIX 4                    17:
   5: 76945    AUDIX 5                    18:
   6: 76946    AUDIX 6                    19:
```

Fig. 3.18

Every Audix is a line to leave a message. We have a server dedicated to save messages, a mailbox. So, we have 6 lines of audix. It means that we can leave six messages simultaneously.

```
display station 76941                                    Page   1 of   3
                              STATION

    Extension: 76941               Lock Messages? n            BCC: 0
        Type: 2500                 Security Code:               TN: 1
        Port: 01A1101              Coverage Path 1:            COR: 1
        Name: AUDIX 1              Coverage Path 2:            COS: 8
                                   Hunt-to Station:          Tests? n


STATION OPTIONS
            Loss Group: 1          Message Waiting Indicator: none
    Off Premises Station? n




            Survivable COR: internal
    Survivable Trunk Dest? y
                                              Remote Office Phone? n
```

Fig. 3.19

We continue with next lines:

```
06 goto step      26              if ani              in     table 18
07 goto step      29              if ani              in     table 19
08 goto step      32              if ani              in     table 20

26 queue-to       skill 134  pri m
27 goto step      10              if unconditionally
28 stop
29 queue-to       skill 135  pri m
30 goto step      10              if unconditionally
31 stop
32 goto vector    34   @step 1  if unconditionally
```

Fig. 3.20

In the line 32 we have a link over vector 34, since vector can contain 32 lines. If we need to add lines, we link with a new vector with 32 lines more. So, in the line 32 we say to the system to go to vector 34, in step 1 (line 1).

```
display vector 34                                       Page   1 of   3
                              CALL VECTOR

    Number: 34                Name: FIT GERMAN #2
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n         Lock? n
    Basic? y     EAS? y   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? y   G3V4 Adv Route? y    CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 queue-to       skill 136  pri m
02 goto vector    21   @step 10 if unconditionally
03 stop
```

Fig. 3.21

The instruction ani means that we have created vrt (vector routing tables) where we can keep a list of phones.

```
list vrt

                    VECTOR ROUTING TABLES

          Number   Name            Sort   Number of Entries

            16     FIT KA HOLLAND    n     0
            17     FIT KA SCANDINA   n     2
            18     FIT KA ASUKANY    n     39
            19     FIT KA THHAHN     n     18
            20     FIT KA KULBRYCH   n     19
```

Fig. 3.22

Then, when the caller comes from this number, we can do something special with this call. In our case, we route it to a specific user, because every user have a list of stores of contact when a clothes are defectives.

```
display vrt 19                                            Page   1 of   3
                      VECTOR ROUTING TABLE

     Number: 19       Name: FIT KA THHAHN        Sort? n

          1: 00498935803107              17: 0049022039128722
          2: 0049931417744               18: 00496173929327
          3: 00496969643159              19:
          4: 0049617392390               20:
          5: 00496173923911              21:
          6: 004961733250824             22:
          7: 00496211672412              23:
          8: 00496211670334              24:
          9: 00496211670145              25:
         10: 00496211672623              26:
         11: 00496211672634              27:
         12: 00496211672214              28:
         13: 004922039128686             29:
         14: 004921115932717             30:
         15: 004969643108                31:
         16: 004921115932723             32:
```

Fig. 3.23

All these lines are managed by a user. When the caller is in the list, the call is routed automatically to the user.

If the system arrives to this line, it means that the call is on queue (waiting to get an agent). Line 9 means that the call is put on queue with low priority (so, if queue contains calls with medium or high priority, they will be reached before)

```
09 queue-to      skill 121  pri l
10 wait-time     10   secs hearing ringback
11 collect       1    digits after announcement 87420    for none
```

If the agents are occupied (speaking with other people or absent), we can hear a ringback and then an announcement:

```
75          87420        integrated     FITGERMANY1STQUEUE
```

After the announcement, the caller can push digit 1 and leave a message. After this the call is hanged up (line 20).

```
12 goto step     19           if digits          =      1
19 messaging     skill 42   for extension active
20 stop
```

3.3.4.   Example of VDN

The Vector Directory Number is used to determine specific properties of the line. It has a general number and in this case it is for FIT Germany. When somebody calls to this number, the VDN is the attendant of this group of calls. All the agents associated to this group can receive the call.

The VDN is then a general number with one origin that has as destiny a group, with a multiple users put in a backup table.

```
display vdn 76321                                    Page   1 of   3
                         VECTOR DIRECTORY NUMBER

                    Extension: 76321
                        Name*: FIT GERMANY
                Vector Number: 21
           Attendant Vectoring? n
           Meet-me Conferencing? n
             Allow VDN Override? y
                          COR: 10
                          TN*: 1
                     Measured: both
    Acceptable Service Level (sec): 10

    VDN of Origin Annc. Extension*:
                     1st Skill*:
                     2nd Skill*:
                     3rd Skill*:




* Follows VDN Override Rules
```

Fig. 3.24

Finally, when we configure the different agents (associated users), we put in a list (for every user) the different groups of callers (VDN's) included.

```
display vdn 76321                                                          Page   2 of   3
                              VECTOR DIRECTORY NUMBER

                                   AUDIX Name: audix

                            Return Destination*:
                    VDN Timed ACW Interval*:
                            BSR Application*:
                   BSR Available Agent Strategy*: EAD-MIA
                             BSR Tie Strategy*: system

                      Observe on Agent Answer? n


                  Display VDN for Route-To DAC*? n
    VDN Override for ISDN Trunk ASAI Messages*? n


                      BSR Local Treatment*? n


* Follows VDN Override Rules
```

Fig. 3.25

4.   Bibliography.

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[7] VoIP:
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http://searchunifiedcommunications.techtarget.com/sDefinition/0,,sid186_gci212388,00.html

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http://www.itu.int/osg/spu/ni/iptel/whatis/index.html

[3] Avaya IP series 9650. Information about the terminal
http://www.avaya.com/gcm/master-usa/en-us/products/offers/9650_ip_telephone.htm&View=ProdDesc

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[5] Avaya Offical Website
http://www.avaya.com/

[6] General information about PBX, Swicth PoE and IP servers
http://www.wikipedia.com/ and other websites (different articles in electronic magazines)

[7] Information about installation of switch PoE to prepare the new network diagram
http://www.extremenetworks.com/products/summit-x250e.aspx

5.   Annex.

Table of buttons recognized by the Avaya server S8700 and used in the project. (extracted from the *Administrator Guide for Avaya Communication Manager*)

Button name	Button label	Description	Maximum
abrv-dial (List: __ DC: __)	AD	Abbreviated Dialing: dials the stored number on the specified abbreviated dialing list. List: specify the list number 1 to 3 where the destination number is stored DC: specify the dial code for the destination number	1 per AD list per dial code
aut-msg-wt (Ext: ___)	Msg (name or ext #)	Automatic Message Waiting: associated status lamp automatically lights when an LWC message has been stored in the system for the associated extension (can be a VDN). This lamp will not light on the mapped-to physical station for messages left for virtual extensions.	1 per aut-mst-ext
after-call Grp:___	AfterCall	After Call Work Mode: allows an agent to temporarily be removed from call distribution in order for the agent to finish ACD-related activities such as completing paperwork. Grp: specify the ACD split group number.	1 per split group
aux-work (RC: __) (Group: __)	AuxWork	Auxiliary Work Mode: removes agent from ACD call distribution in order to complete non-ACD-related activities. RC: Optional assignment for the 1- or 2-digit Reason Code to be used to change to Aux Work using this button, when Reason Codes is active. Multiple Aux Work buttons, each with a different RC, can be assigned to the same station set. Grp: The split group number for ACD.	1 per split group
auto-in (Group: __)	Auto In	Auto-In Mode: allows the user to become automatically available for new ACD calls upon completion of an ACD call. Grp: The split group number for ACD.	1 per split group

Button name	Button label	Description	Maximum
call-appr	extension	Call Appearance: originates or receives calls. Assign to a 2-lamp appearance button.	Depends on station type
check-out	Check Out	Check Out (*display button*): Changes the state of the associated guest room to vacant and turns on the outward calling restriction for the guest room's station. Also clears (removes) any wake-up request for the station.	1 per station
check-in	Check In	Check In (*display button*): changes the state of the associated guest room to occupied and turns off the outward calling restriction for the guest room's station.	1 per station
q-calls (Grp: ___)	QueueCall	Queue Calls: associated status lamp flashes if a call warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station

Switch PoE.

A network switch is a computer networking device that connects network segments.

Low-end network switches appear nearly identical to network hubs, but a switch contains more "intelligence" (and comes with a correspondingly slightly higher price tag) than a network hub. Network switches are capable of inspecting data packets as they are received, determining the source and destination device of that packet, and forwarding it appropriately. By delivering each message only to the connected device it was intended for, a network switch conserves network bandwidth and offers generally better performance than a hub.

In the past, it was faster to use Layer 2 techniques to switch, when only MAC addresses could be looked up in content addressable memory (CAM). With the advent of ternary CAM (TCAM), it was equally fast to look up an IP address or a MAC address. TCAM is expensive, but very appropriate for enterprise switches that use default routes plus a moderate number of other routes. For routers that need a full Internet routing table, TCAM may not be cost-effective.

As with hubs, Ethernet implementations of network switches support either 10/100 Mbit/s or 10/100/1000 Mbit/s ports Ethernet standards. Large switches may have 10 Gbit/s ports. Switches differ from hubs in that they can have ports of different speed.

The network switch, packet switch (or just switch) plays an integral part in most Ethernet local area networks or LANs. Mid-to-large sized LANs contain a number of linked managed switches. Small Office, Home Office (SOHO) applications typically use a single switch, or an all-purpose converged device

such as gateway access to small office/home office broadband services such as DSL router or cable, WiFi router. In most of these cases, the end user device contains a router and components that interface to the particular physical broadband technology, as in the Linksys 8-port and 48-port devices. User devices may also include a telephone interface to VoIP.

Network switch is a marketing term rather than a technical one. Switches may operate at one or more OSI layers, including physical, data link, network, or transport (i.e., end-to-end). A device that operates simultaneously at more than one of these layers is called a multilayer switch, although use of the term is diminishing.

In switches intended for commercial use, built-in or modular interfaces makes it possible to connect different types of networks, for example Ethernet, Fibre Channel, ATM, and 802.11. This connectivity can be at any of the layers mentioned. While Layer 2 functionality is adequate for speed-shifting within one technology, interconnecting technologies such as Ethernet and token ring are easier at Layer 3.

Again, "switch" is principally a marketing term; interconnection of different Layer 3 networks is done by routers. If there are any features that characterize "Layer-3 switches" as opposed to general-purpose routers, it tends to be that they are optimized, in larger switches, for high-density Ethernet connectivity.

In some service provider and other environments where there is a need for much analysis of network performance and security, switches may be connected between WAN routers as places for analytic modules. Some vendors provide firewall, network intrusion detection, and performance analysis modules that can plug into switch ports. Some of these functions may be on combined modules.

In other cases, the switch is used to create a "mirror" image of data that can go to an external device. Since most switch port mirroring provides only one mirrored stream, network hubs can be useful for fanning out data to several read-only analyzers, such as intrusion detection systems and packet sniffers.

Layer-specific functionality

A modular network switch with 3 network modules (a total of 24 Ethernet and 14 Fast Ethernet ports) and one power supply.

While switches may learn about topologies at many layers, and forward at one or more layers, they do tend to have common features. Other than for computer-room very high performance applications, modern commercial switches use primarily Ethernet interfaces, which can have different input and output speeds of 10, 100, 1000 or 10000 megabits per second. Switch ports almost always default to full-duplex operation, unless there is a requirement for interoperability with devices that are strictly half duplex. Half-duplex means that

the device can only send or receive at any given time, whereas full-duplex can send and receive at the same time.

At any layer, a modern switch may implement power over Ethernet (PoE), which avoids the need for attached devices, such as an IP telephone or wireless access point, to have a separate power supply. Since switches can have redundant power circuits connected to uninterruptible power supplies, the connected device can continue operating even when regular office power fails.

Layer-1 hubs versus higher-layer switches

A network hub, or repeater, is a fairly unsophisticated broadcast device, and rapidly becoming obsolete. Hubs do not manage any of the traffic that comes through them. Any packet entering a port is broadcast out or "repeated" on every other port, save the port of entry. Since every packet is repeated on every other port, packet collisions result, which slows down the network.

Hubs have actually become hard to find, due to the widespread use of switches. There are specialized applications where a hub can be useful, such as copying traffic to multiple network sensors. High end switches have a feature which does the same thing called port mirroring. There is no longer any significant price difference between a hub and a low-end switch.

Layer 2

A network bridge, operating at the Media Access Control (MAC) sublayer of the data link layer, may interconnect a small number of devices in a home or office. This is a trivial case of bridging, in which the bridge learns the MAC address of each connected device. Single bridges also can provide extremely high performance in specialized applications such as storage area networks.

Bridges may also interconnect using a spanning tree protocol that allows the best path to be found within the constraint that it is a tree. In contrast to routers, bridges must have topologies with only one active path between two points. The older IEEE 802.1D spanning tree protocol could be quite slow, with forwarding stopping for 30-90 seconds while the spanning tree would reconverge. A Rapid Spanning Tree Protocol was introduced as IEEE 802.1w, but the newest edition of IEEE 802.1D-2004, adopts the 802.1w extensions as the base standard.

While "layer 2 switch" remains more of a marketing than a technical term, the products that were introduced as "switches" tended to use microsegmentation and full duplex to prevent collisions among devices connected to Ethernets. By using an internal Forwarding Plane much faster than any interface, they give the impression of simultaneous paths among multiple devices.

Once a bridge learns the topology through a spanning tree protocol, it forwards data link layer frames using a layer 2 forwarding method. There are four forwarding methods a bridge can use, of which the second through fourth method were performance-increasing methods when used on "switch" products with the same input and output port speeds:

> 1. Store and forward : The switch buffers and, typically, performs a checksum on each frame before forwarding it on.

2. Cut through : The switch reads only up to the frame's hardware address before starting to forward it. There is no error checking with this method.
3. Fragment free : A method that attempts to retain the benefits of both "store and forward" and "cut through". Fragment free checks the first 64 bytes of the frame, where addressing information is stored. This way the frame will always reach its intended destination. Error checking of the actual data in the packet is left for the end device in Layer 3 or Layer 4 (OSI), typically a router.
4. Adaptive switching : A method of automatically switching between the other three modes.

Note that cut-through switches have to fall back to store and forward if the outgoing port is busy at the time the packet arrives. While there are specialized applications, such as storage area networks, where the input and output interfaces are the same speed, this is rarely the case in general LAN applications. In LANs, a switch used for end user access typically concentrates lower speed (e.g., 10/100 Mbps) into a higher speed (at least 1 Gbps). Alternatively, a switch that provides access to server ports usually connects to them at a much higher speed than is used by end user devices.

Layer 3

Router is a marketing term for a Layer 3 switch, typically a router optimized for Ethernet interfaces. Like other switches, it connects devices to single ports for microsegmentation. The ports normally operate in full duplex.

Switches, even primarily Layer 2 switches, can be aware of Layer 3 multicast and increase efficiency by delivering the traffic of a multicast group only to ports where the attached device has signaled that it wants to listen to that group. In a switch not aware of multicasting and broadcasting, frames are also forwarded on all ports of each broadcast domain, but in the case of IP multicast this causes inefficient use of bandwidth. To work around this problem some switches implement IGMP snooping.

Layer 4

While the exact meaning of the term Layer-4 switch is vendor dependent, it almost always starts with a capability for network address translation, but then adds some type of load distribution based on TCP sessions.

The device may include a stateful firewall, a VPN concentrator, or be an IPSec security gateway.

Layer 7

As with the other types of switches, Layer 7 is a marketing term. They may distribute loads based on URL or by some installation-specific technique to

recognize application-level transactions. A Layer-7 switch may include a web cache and participate in a content delivery network.

A rack-mounted switch with network cables

Managed switches — These are ones which allow access to one or more interfaces for the purpose of configuration or management of features such as Spanning Tree Protocol, Port Speed, VLANs, etc. High-end or "enterprise" switches may provide a serial console and command-line access via telnet and ssh, as well as management via SNMP. More recent devices may also provide a web interface. Limited functions, such as a complete reset by pushing buttons on the switch are usually also provided. Managed switches are found in medium or large "enterprise" networks and though more expensive are of higher quality (e.g. with a backplane with higher transfer speeds). The task of managing usually requires understanding of Layer 2 networks (e.g. Ethernet).

Smart (or intelligent) switches — These are managed switches with a limited set of features. Likewise "web-managed" switches are switches which fall in a market niche between unmanaged and managed. For a price much lower than a fully managed switch they provide a web interface (and usually no CLI access) and allow configuration of basic settings, such as VLANs, port-speed and duplex.

Web-managed switches — Similar in functionality to a smart switch. A Web-managed switch is configured through a browser instead of via a desktop utility.
Traffic monitoring on a switched network

Unless port mirroring or other methods such as RMON or SMON are implemented in a switch, it is difficult to monitor traffic that is bridged using a switch because all ports are isolated until one transmits data, and even then only the sending and receiving ports can see the traffic. These monitoring features rarely are present on consumer-grade switches.

Two popular methods that are specifically designed to allow a network analyst to monitor traffic are:

Port mirroring — the switch sends a copy of network packets to a monitoring network connection.

SMON — "Switch Monitoring" is described by RFC 2613 and is a protocol for controlling facilities such as port mirroring.

Another method to monitor may be to connect a Layer-1 hub between the monitored device and its switch port. This will induce minor delay, but will

provide multiple interfaces that can be used to monitor the individual switch port.

Typical switch management features

(In order of basic to advanced):

Turn some particular port range on or off

Link speed and duplex settings

Priority settings for ports

MAC filtering — and other types of "port security" features which prevent MAC flooding

Use of Spanning Tree Protocol

SNMP monitoring of device and link health

Port mirroring (also named: port monitoring, spanning port, SPAN port, roving analysis port, link mode port)

Link aggregation (also called: bonding/trunking)

VLAN settings 802.1X network access control

Link aggregation allows you to use multiple ports for the same connection achieving higher data transfer speeds. Creating VLANs can serve security and performance goals by reducing the size of the broadcast domain.

Managed or Unmanaged Switching

This option determines the level of control you can have over your network, including the degree of security that your business wants for its networked data and communications. Managed switches offer more control; they also usually require more technical expertise. If you have multiple LANs and switches, you may decide to manage some of them and not others.

Unmanaged switches are preconfigured to satisfy most small business needs and can be quickly set up, making them appropriate for simple LAN connections.

Managed switches provide control capabilities that can increase LAN security and performance, and let you segment traffic. For example, you could allow only certain employees access to your finance applications. You could also give certain applications more bandwidth, or ensure that video and voice communications take priority over other types of traffic.