Facet*Phone*™

System Administration Guide

Facet*Corp*

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Multiple Instances of FacetPhone on a Server

Introduction

The FacetPhone System Administration Guide contains all documentation of relevance to the system administrator including instructions for installation and configuration of the system.

FacetPhone Concepts for System Administrators.

There are a few basic FacetPhone concepts that you need to understand to administer it properly.

User extension numbers vs. telephone station numbers.

FacetPhone has the concept of roving "user extensions." This allows users' calls to be directed to different telephones depending on where they are. For example, a user may primarily use the phone on their desk at the office. However, they may also have a FacetPhone IP phone or gateway at home with a connection to the office via a high speed Internet service such as DSL or cable. When they login to FacetPhone and set their location to the office, their calls are directed to the phone on their desk. When they login at home and set their location to home, their calls are directed to their phone at home.

Therefore, a user's extension number and a telephone station number are two different things. A user's phone at their office and their phone at home have different station numbers. However, their extension number is always the same, and callers can reach them by dialing their extension regardless of the particular telephone station that they are currently using.

Location profiles.

Location profiles make the connection between a user and his phone at a particular location. For example, a user's "office" location profile would associate the user with the phone is his office and a "home" location profile would associate the user with telecommuter phone at home. In addition, all users have a "not logged in" profile that specifies which phone, if any, should be associated with the user when they are not logged in to the FacetPhone system. In addition to phone association, location profiles contain other settings which it is useful to maintain on a per-location basis.

User groups.

FacetPhone has the concept of user groups. In a small business, there may only be one group which is usually given the company name. Additional groups may be defined such as sales, support, etc. Phones and lines are assigned to a group and controlled by the operators for that group. Auto-attendants are selected and scheduled independently for each group. Groups may also be defined to control which operators or backup operators are rung from an auto-attendant. A group has voice mail folders and can therefore be used to take voice mail that multiple people can access. The company directory is also based on the members of the group that is currently selected when the auto-attendant is running the company directory.

System Architecture

Overview

The FacetPhone system consists of

- A computer running the FacetPhone server software.

- IP phones which serve as the telephone instruments for most users.

- IP telephony gateways that connect analog telephone instruments and telephone lines from the phone company to your LAN or WAN.

- Desktop PCs running the FacetPhone client software including the user interface, the Windows TAPI service provider and Caller ID notification program.

By providing remote users and branch offices with a VPN across a fast Internet or private MPLS connection, a FacetPhone system may be distributed geographically.

System Diagram

The following is the topology of a FacetPhone system:
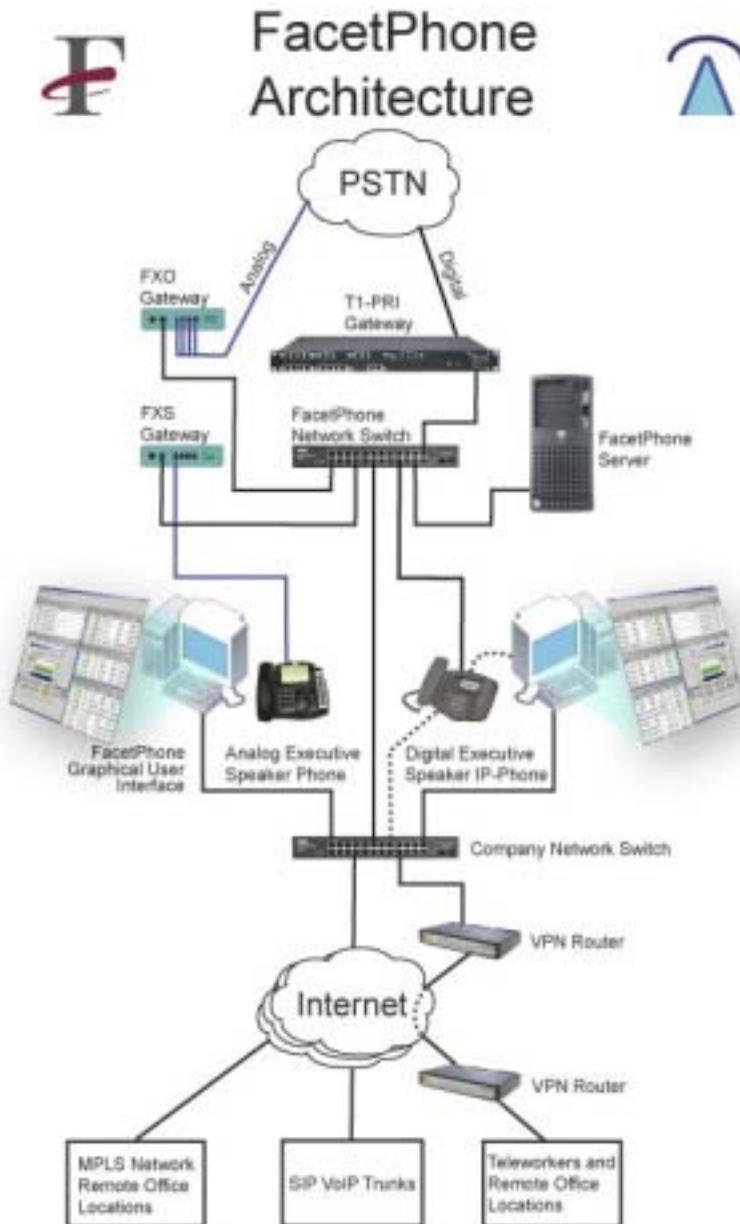
Server Software

The FacetPhone server software includes the following programs:

fp_db

> The database program manages all the data that is needed by the other programs. The database receives information from user interfaces and the PBX and then distributes this information to other user interfaces and the PBX.

fp_pbx

> The PBX program operates the gateways and performs all of the call control functions. It spawns IVR processes (fp_ivr) when a caller is put into an auto-attendant, voice mail or music/message on hold.

fp_ivr

> The IVR program (Interactive Voice Response) performs all of the auto-attendant and voice mail functions. Whenever a caller is listening to recorded prompts, operating a touch tone menu or recording messages, it is the fp_ivr program which provides this interaction. The fp_ivr program is controlled by IVR scripts which specify the operation of a particular auto-attendant. The IVR scripts are written in a simplified, C-like language. A separate fp_ivr process is run for each phone or line that is in voice mail, auto-attendant, or music/message on hold.

fp_conf

> The conference program mediates conference calls. All of the RTP audio packets are sent to the fp_conf process where all inputs are mixed and then retransmitted to all parties in the conference. A separate fp_conf process is run for each conference.

fp_client

> The client program is a utility that is run on the server to send commands to the database as though a user was performing these functions from a user interface.

fp_telnet

> The fp_telnet program is a special purpose telnet program that is used to program gateway settings on gateways that support telnet configuration.

fp_tftp

> The fp_tftp program is a special purpose TFTP program that is used to load new software in gateways that support TFTP.

fp_smtp

> The fp_smtp program is a special purpose SMTP program that is used to email voice mail and voice mail notifications to your main email server.

fp_vconv

> The fp_vconv program converts voice mail to the appropriate format and volume for playing on the PC speakers and for sending as an email attachment. It also converts all voice mail and prompts to the uncompressed or compressed format depending on which way the audio was originally recorded.

fp_run

> The fp_run program is a special purpose program that the system can use to run any program or shell script asynchronously and have its success or failure reported back to the parent.

fp_mysqldb

> The fp_mysqldb program reads call detail records and user status records from the fp_db program and writes them into a MySQL database.

fp_aagen

> The fp_aagen program is used to generate auto-attendant IVR scripts from auto-attendant configuration files.

fp_rtpnat

> The fp_rtpnat program is used to route voice packets through the server instead of directly between the endpoints. This is necessary for support of some SIP trunking services.

facetphone

> The facetphone shell script is used to start and stop the FacetPhone servers, signal the fp_db program to reread the configuration, restart the system, take over backup operation, or resume master operation.

Client Software

The FacetPhone client software includes the following programs:

User Interface.

> The user interface is a Java application. Any PC on which it is run must have the Java Runtime, Version 1.5 or later installed. An appropriate JRE installation for Windows is included with the software and the latest version for any platform can always be downloaded from www.java.com.

TAPI Service Provider.

> The TAPI service provider is only installed on Windows PCs. It interfaces the FacetPhone system to the Windows TAPI server which allows TAPI enabled applications like Outlook to dial. The FacetPhone TAPI service provider also allows delivery of caller ID data to TAPI enabled programs that make use of this such as Act!

Caller ID Program.

> The Caller ID program is only installed on Windows PCs. It provides FacetPhone incoming caller ID data to specific applications such as Outlook and Goldmine.

IP Phones and Soft Phones

An IP phone is a telephone instrument and VoIP gateway combined into one. The IP phone plugs directly into your network. FacetPhone currently supports only MGCP IP phones.

A soft phone is software which runs on your PC and uses the sound card and a microphone and speaker as an IP telephone. Use of a PC headset or special USB phones can make the soft phone a convenient telephone instrument. Remember, however, that when your PC is down, your phone does not work. FacetPhone currently supports only MGCP soft phones.

The IP phones and soft phones currently supported are:

Polycom PC301, PC501, PC601. The function keys and display are extensively utilized by FacetPhone to make FacetPhone's features easily accessible.

Cisco 7940 and 7960 series phones. The display and soft keys are extensively utilized by FacetPhone.

eyeP Media's MGCP eyeP Phone. This soft phone may be purchased from FacetCorp.

IP Telephony Gateways

IP telephony gateways connect analog telephone instruments and analog lines to your computer network. Each port on a gateway is either FXS, which connects to an analog telephone station, or FXO, which connects to an analog outside phone line from your telephone company. In addition, a T1 gateway is used to connect digital phone service from the phone company to your network.

Gateways currently supported are:

Voipack VOG4000 4 port gateway. Each port is individually configurable as FXS or FXO.

Cisco ATA 186 2 port FXS gateway.

Adtran Total Access 612/616/624 12-24 port FXS gateway.

Grandstream GXW410x FXO gateway available in 4 and 8 port configurations.

Audiocodes Mediant 1000 T1 gateway.

Installation and Configuration

Installation Plan

The following tasks will be performed to install and setup the FacetPhone system:

- Information about the site will be collected including all user information and desired extension numbers. FacetCorp will supply you with a site survey to be used in specifying all the information needed for the installation. A smooth installation depends on filling out this survey completely.

- The FacetPhone server's operating system must be installed and configured properly so that the necessary OS services are installed.

- The FacetPhone server software will be installed.

- The IP phones will be configured.

- The gateways will be configured.

- The FacetPhone system will be configured according to the information gathered about the site.

- Auto-attendant scripts and prompts will be customized for the site.

- Any backup servers included in the system will be setup.

- The FacetPhone User Interface will be installed on the PCs that will access FacetPhone. Optionally the FacetPhone TAPI and Caller ID software will be installed on PCs which will be used with TAPI enabled Windows software.

Gathering information about the site

Before beginning an installation, get an up to date site survey document from FacetCorp. Most of the information requested on the pre-installation site survey is self-explanatory. However, the following subjects should be understood before filling out the survey.

Defining the numbering plan

A site's numbering plan determines what the user's extension numbers will be, what the telephone station numbers will be, what line numbers will be and what digits are available for auto-attendant menu choices. All of these considerations must be taken into account when establishing the numbering plan. FacetPhone is very flexible in its numbering scheme. However, there are certain guidelines to follow to ensure that the numbering plan will make sense to the users and allow a fully functional auto-attendant.

User extension numbers and telephone station numbers must all contain the same number of digits. For example, at a site that will have a maximum of 100 users, you cannot number extensions 1 through 100. Some users would have a single digit extension, most would have two digit extensions and the person assigned extension 100 would have a three digit extension. In this case, it would be best to number the user and phone extensions from 100 to 199.

In general, for sites of 100 users or less, we recommend three digit user extension and telephone station numbers beginning with 100. For sites of 1000 users or less, you should use four digit numbers starting with 1000, etc. Sites with less than 10 can use two digit extension and station numbers, but it is best to size the numbering plan for the largest size the site is likely to become since users will find it disruptive if a numbering plan change is required in the future.

The other consideration in defining the numbering plan is speed dial numbers. FacetPhone allows you to define speed dial numbers of any range as long as a speed dial number does not start with the same digit as a user extension or station number. By default, the two digit speed dial numbers 20 - 59 are reserved for the definition of 40 speed dial numbers.

Assigning extension and station numbers

Once the numbering plan is established, the user extension and telephone station numbers can be assigned. Because FacetPhone allows users' extensions to "roam", allowing them to have their calls directed to any station, there are two different sets of numbers to assign, user extensions and telephone stations. If you are replacing an existing system, you will probably want to assign the current station numbers to the new stations. If your current system uses two digit station numbers and you are using three digit numbers in your FacetPhone numbering plan, then just add 1 to the beginning of the old numbers.

User extension numbers should be assigned based on the station the person normally uses. If the phone in your office is station 120, then you should make your user extension number 120.

Assigning line numbers

Line numbers are usually numbered from 1 through the number of lines you have. Make a note of which phone numbers are assigned to which line numbers. If you have multiple lines on a rotary from the phone company, you probably want to make the first line on your rotary be line 1, the next one line 2 and so on throughout your rotary order.

Identifying groups

The FacetPhone installation procedure will create a default group for you that includes all users and is the default operator group for all phones and lines.

In addition, you might want to define other, smaller groups of people. FacetPhone allows you to watch the status of people in a group, so you might find it useful to watch a particular group of people to see who is available. For example, at FacetCorp we have Sales and Support groups defined. Our operators can look at these smaller group displays to more easily see which sales or support person is available to take a call.

In larger companies you may have DID numbers routed to different groups of operators. Each group has independently selected and scheduled auto-attendants.

Server Preparation

It is highly recommended that the server and all gateways be powered through an Uninterruptible Power Supply (UPS), so that your phone system will be unaffected by a brief power outage.

As of this version, FacetCorp is using the Centos 5.2 version of Linux on the servers it prepares. Generally, FacetCorp will install the operating system on most FacetPhone servers. However, if you are doing this yourself, please contact FacetCorp for detailed instructions for installation of the operating system.

FacetPhone Server Software Installation

The latest FacetPhone software is available to anyone under FacetPhone software maintenance by contacting FacetCorp support. The software will come as a CPIO archive.

Login as root on the FacetPhone server. Put the FacetPhone CPIO file in a temporary directory on the FacetPhone server. Extract the installation files from the archive with the following command depending on your operating system:

> Linux:
>
> > cpio -idumv < facetphone.cpio
>
> SCO, AIX (and most others):
>
> > cpio -icdumv < facetphone.cpio

This should create the files:

> facetphone.z
> install.sh
> partno.txt
> PC_CLIENT/setup.exe
> PC_CLIENT/FP_TSUI.exe
> PC_CLIENT/FP_Utapi.exe
> fp_utapi.z
> utapi_install.sh

You can delete the cpio file after its contents have been successfully extracted.

Before you run the installation procedure, you should have the following information available to answer questions asked during the installation:

The top level directory where the web pages are kept on your FacetPhone server. This is operating system dependent. The default value for your operating system will be provided.

The port number used by your web server if it is different from the default port 80.

The directory where you want FacetPhone installed. The default is /usr/facetphone. If you want to relocate it due to disk space considerations (all voice mail will be stored under this directory), then the installation procedure will give you the opportunity to specify an alternate directory.

The IP address of your FacetPhone server. If the server has multiple IP addresses, use the one that the gateways can use to send messages to the server.

The home directory for the TFTP server. This is operating system dependent.

The name you want to give the default group that is automatically created at installation time. This is usually your company name.

The IP address of the mail server that is used on your network.

The port number used by your mail server if it is different from the default port 25.

The method of dialing local calls at this location, 10 digit dialing, 7 digit dialing or both.

Having this information available will assure the smoothest possible installation. However, if you need to change one of the answers later, it is easy to do.

Run the installation procedure with the command:

./install.sh

The installation procedure will run. If it is successful and you are upgrading an existing installation, the FacetPhone servers will be started when the installation procedure

completes. If it is a first time installation, the servers will not be automatically started. You can start the FacetPhone servers with the command:

```
facetphone start
```

If the installation fails, please call FacetCorp support for help.

Configuring the VoIP Gateways and IP Phones

The following sections discuss each of the gateways and IP phones that are supported by FacetPhone. Generally, each of these devices will be shipped to you preconfigured and you should only need to plug them into your network and turn them on. However, the basic configuration procedures are given in the following sections to facilitate on-site changes. If this is the first time you have configured a particular type of device, please consult with FacetCorp support staff before beginning.

Voipack Gateway Setup

The Voipack VOG4000 is a 4 port gateway controlled with the MGCP protocol. Each port is independently configurable as FXS to attach a phone or FXO to attach an analog line from the phone company. Analog lines must be "loop start". "Ground start" lines are not supported.

Unpack the gateway and set it in the location you have chosen that has access to power, the LAN, the telephone lines from the phone company and/or the extension lines to analog phones that will be used.

Attach the gateway to the LAN with the cable provided. Plug the power supply into the gateway and into a power strip. You should put your gateways on an Uninterruptible Power Supply so the phone system will stay up during a power outage.

With a Voipack gateway, the only configuration that must be done outside of the FacetPhone administrative interface is setting its IP address and the address of a routing gateway if necessary. The gateway may have been shipped to you with its IP address preset if this information was known. If your gateway has a label on it indicating the IP address that you want, then try pinging the gateway:

> ping xxx.xxx.xxx.xxx

where you provide the IP address that is on the label. If the ping succeeds, then you are finished with this part of the gateway setup.

If you need to change the gateway's IP address, follow one of the procedures below. The method using telnet is probably easier if you know what the gateway's current IP address is. Otherwise, you can use the serial port method if you do not know the gateway's current IP address.

Once you have set the IP address, ping the gateway from the FacetPhone server to verify connectivity. All other Voipack gateway configuration will be done from the FacetPhone administrative interface.

Setting the IP Address on a Voipack Gateway Using Telnet

If the gateway was not shipped with the IP address preset to your desired value, then its initial IP address will be 192.168.37.37.

If you are not able to ping the gateway, then proceed to the next section, "Setting the IP Address on a Voipack Gateway Using its Serial Port".

Once you can ping the gateway, then telnet to it with the command:

    telnet 192.168.37.37

The gateway should give the prompt:

    Voipack>

Change the IP address to the desired value with the following command to the gateway:

    ipAddrSet "xxx.xxx.xxx.xxx"

where the desired address is supplied. Be sure to enclose the address in double quotes. You can verify the value of the setting with the command:

    ipAddrGet

If the gateway needs to communicate with a server or other gateways on a different network, then the IP address of the routing gateway must be provided:

    gwAddrSet "xxx.xxx.xxx.xxx"

You can verify the value of the routing gateway address with the command:

    gwAddrGet

After setting the IP address, reboot the gateway with the command:

    reboot

All further configuration of the Voipack gateway on the FacetPhone system will be done from the FacetPhone administrative interface as described in the "Defining Gateways" section below.

Setting the IP Address on a Voipack Gateway Using its Serial Port

You can use any method that is most convenient for you to connect to the serial port on the gateway. The instructions below are for using HyperTerminal from a Windows PC.

Attach a standard DB-9 to DB-9 cable from COM 1 of a PC to the serial port on the gateway.

Use any terminal emulation program that will let you communicate with COM 1.

HyperTerminal is on most Windows PCs:

> Start->Programs->Accessories->Communications->HyperTerminal

Or on some systems:

> Start->Programs->Accessories->HyperTerminal->HyperTerminal

Double click hypertrm.exe

Give your configuration a name like Voipack and click OK.

On the "connect using" screen, leave the setting at COM1 or change to the COM port you are using and press OK.

On the configuration screen, change to 9600 baud, 8 bit, no parity, 1 stop bit, and flow control none or xon-xoff.

Once you have a good serial connection to the gateway it will give you the prompt:

Voipack>

If it does not, power the gateway off and back on. It should display a boot up sequence followed by the Voipack> prompt.

Change the IP address to the desired value with the following command to the gateway:

> ipAddrSet "xxx.xxx.xxx.xxx"

where the desired address is supplied. Be sure to enclose the address in double quotes. You can verify the value of the setting with the command:

> ipAddrGet

If the gateway needs to communicate with a server or other gateways on a different network, then the IP address of the routing gateway must be provided:

> gwAddrSet "xxx.xxx.xxx.xxx"

You can verify the value of the routing gateway address with the command:

gwAddrGet

After setting the IP address, reboot the gateway with the command:

reboot

All further configuration of the Voipack gateway on the FacetPhone system will be done from the FacetPhone administrative interface as described in the "Defining Gateways" section below.

Reconfiguring the FXS and FXO ports on a Voipack Gateway

The ports on a Voipack gateway may be configured as either FXS or FXO port. Analog telephone instruments are plugged into FXS ports and analog lines from the phone company are plugged into FXO ports. Some devices such as an overhead paging system may require either FXS or FXO and the documentation for these devices must be consulted to determine which they require. **Reconfiguration of the ports on a Voipack gateway should only be done with the supervision of a FacetCorp support engineer.**

FacetPhone requires that all the FXS ports on a gateway come before the FXO ports. For example, if a gateway will have 2 FXS ports and 2 FXO ports, then ports 1 and 2 must be FXS and ports 3 and 4 FXO.

There is a jumper and a dip switch for each port. The following diagram shows the position of the jumpers and dip switches for FXS ports 1 and 2 and FXO ports 3 and 4. The front of the gateway with its cover off would be at the bottom of the diagram.
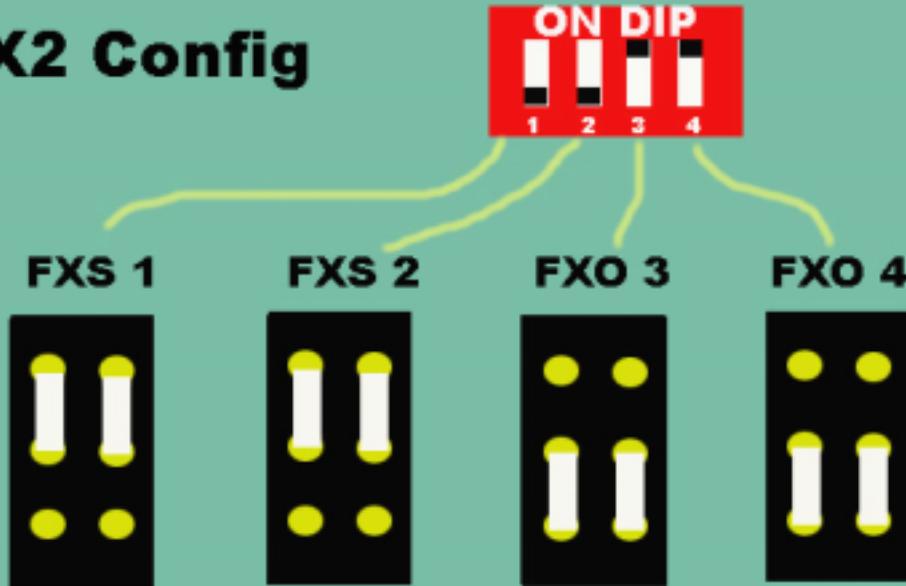
After configuring the jumper and switch properly for each port, power up the gateway and telnet to it and enter the following command:

        Voipack> flash clean

Then power off the gateway before giving any other commands and restart it. Finally be sure that the FacetPhone system is properly configured for the number of FXS and FXO ports on the gateway and then send settings to the gateway. See the section Defining Gateways later in this manual.

Cisco ATA 18x Gateway Setup

The Cisco ATA 18x gateway is a 2 port FXS gateway. Only one port can use the G729 codec, so in applications where this codec will be used, the gateway is configured in the FacetPhone system as a 1 port gateway. FacetPhone operates the gateway using the MGCP protocol.

The Cisco ATA's initial IP address is configured with the phone menu. All other settings for this gateway are currently administered through the gateway's web page.

Preparing the Gateway

Plug a phone into the Phone 1 jack on the gateway, plug in the network cable and plug in the power jack.

Setting the IP Address

Pick up the phone and press the button on the top of the gateway. A menu will be presented over the phone. Press 1# to set the IP address. Enter the address using the * key for the dots and terminate the entry with the # key. For example:

        192*168*1*200#

Another menu will be presented. Select 3 to save the IP address. Hang up the phone and pick it back up, press the button on the top of the gateway, and press 21# to verify the new address. Finally, press menu option 20# to turn off DHCP.

The gateway should now be reachable on the network. Verify with a ping command from a UNIX or DOS prompt:

        ping 192.168.1.200

Configuring the Gateway Settings

Cycle power on the gateway. With a web browser, go to the IP address of the gateway /dev. For example:

        http://192.168.1.200/dev

You should set or verify the setting of the following parameters on this form:

UseTftp:

>0

TftpURL:

>IP address of the FacetPhone server (for future use).

CfgInterval:

>31536000

Dhcp:

>0

StaticIP:

>Should already be set to the value that was entered from the telephone menu.

StaticRoute:

>The default route that the gateway should use to reach the FacetPhone server and all other gateways. If the entire FacetPhone system is on the same LAN, this field may be left blank.

StaticNetMask:

>As defined by your network, probably 255.255.255.0

CA0orCM0:

>IP address of your FacetPhone server followed by :2727

PrfCodec:

>1

ConnectMode:

>0x10000404

AudioMode:

0x00310031

Domain:

IP address of the gateway within square brackets, such as [192.168.1.200].

After you have the correct values in the form, click the Apply button to store the new values in the gateway.

If you ever have a need to load new firmware in your Cisco gateway, please contact FacetCorp support, as this procedure varies depending on the version of the gateway and the current version of the firmware.

All further configuration of the Cisco gateway on the FacetPhone system will be done from the FacetPhone administrative interface as described in the "Defining Gateways" section below. Note that the AudioMode entered above will require you to check the box for "RFC2833 digit detection" on the Gateway configuration screen in the Facet-Phone administrative interface.

Adtran Total Access 612/616/624 FXS Gateway Setup

The Adtran Total Access FXS gateway comes in configurations of 12, 16 or 24 ports. This gateway does not support the G729 codec. FacetPhone operates the Adtran gateway using the MGCP protocol.

The configuration of this gateway is sufficiently complex that it is only done by Facet-Corp support engineers.

Audiocodes Mediant 1000 T1 Gateway Setup

The Audiocodes Mediant 1000 connects 1 to 4 T1s to the LAN. Many T1 protocols are supported on the PSTN side of the connection. FacetPhone operates this gateway using the SIP protocol.

The configuration of this gateway is sufficiently complex that it is only done by Facet-Corp support engineers.

Polycom 301/501/601 IP Phone Setup

The Polycom 301, 501 and 601 IP phones are supported by FacetPhone including the display which is updated with informative FacetPhone messages when the phone is in use on a FacetPhone system. FacetPhone operates these phones with the MGCP protocol.

To perform the basic setup on the Polycom phone, reboot the phone by simultaneously holding down the VOL-, VOL+, Messages and Hold keys. Once the display has a SETUP softkey displayed and is counting down the seconds until reboot, press the SETUP softkey.

Enter the password which is 456 and press the OK softkey.

If the DHCP client is enabled, disable it by pressing the EDIT softkey then the right arrow key until the DHCP Client setting is Disabled and then press the OK softkey.

Press the down arrow key to highlight the Phone IP Addr item. To set the IP address for the phone, press the EDIT softkey, and enter the IP address using the * key to enter a dot in the IP address. The Del key may be pressed to backspace. When the address is entered correctly, press the OK softkey.

Press the down arrow key to highlight the Subnet Mask. Press the EDIT softkey and enter the appropriate subnet mask such as 255.255.255.0. The Del key will backspace and the * key is used to enter a dot the same as when entering the phone's IP address. Press the OK softkey when the subnet mask is entered properly.

Press the down arrow key to highlight the IP Gateway. This is the gateway that is used to route packets from the phone to other subnets on the LAN. If you need to enter this address, it is done the same as the other address entries.

Press the down arrow key to highlight the Server Menu item. Press the SELECT softkey and the server items will be displayed. The Server Type should be set to FTP. If it isn't, press the EDIT softkey and use the right arrow key to change the setting until it is FTP and then press the OK softkey.

Set the Server Address to be the IP address of the FacetPhone server using the same procedure to enter the other IP addresses above. When done, press the EXIT softkey to return to the previous menu.

Press the down arrow key to highlight the SNTP Address. Press the EDIT softkey to change the value. Press the softkey labeled "a->1A" to get "1->Aa" for numeric input. Then enter the FacetPhone server address the same way you have entered the other IP addresses. Press the OK softkey when the address is entered correctly.

Press the down arrow key to highlight the GMT offset. Press the EDIT softkey to change the value. Use the left and right arrow keys until the proper value is displayed (eg. -5 for Eastern time zone, -6 for Central, etc.). Press the OK softkey when the value is correct.

Press the down arrow key until the CDP item is highlighted. Press the EDIT softkey and then press the right arrow key until the value is Disabled. Press the OK softkey.

Press the EXIT softkey. If you have made any changes to the configuration, the Exit Option menu will be presented. Use the up or down arrow keys to highlight the Save & Reboot option and then press the SELECT softkey. The phone will upload a log file to the FacetPhone server and will then reboot.

Cisco 7940/7960 IP Phone Setup

The Cisco 7940 and 7960 IP phones are supported by FacetPhone including the display which is updated with informative FacetPhone messages when the phone is in use on a FacetPhone system. FacetPhone operates these phones with the MGCP protocol.

To configure the Cisco IP phone, plug in the power cord and press the Settings button that shows a check mark in a box. Use the down arrow button to highlight the Unlock Config item. Press the Select softkey and it will prompt for a password. The phone must be in Alpha entry mode. If there is a Number softkey, then it is in Alpha mode. If there is an Alpha softkey, then press it to put the phone into Alpha entry mode. Then enter the password which is "cisco" by pressing each key multiple times until the correct letter is displayed, wait until the letter is entered and then proceed to enter the next letter. For example to enter "cisco" press the 2 key three times to enter "c" then pause, press the 4 key three times to enter "i" and pause, press the 7 key four times to enter "s" and pause, press the 2 key three times to enter "c" and pause, and finally press the 6 key three times to enter the "o". Then press the Accept key to submit the password. The configuration should now be unlocked and the Unlock Config item should now be changed to Lock Config.

Use the up arrow to highlight the Network Configuration item and press the Select softkey. Then use the down arrow key to highlight the IP Address item and press the Edit softkey. The left most softkey allows you to backspace and the second from the left softkey allows you to toggle between Alpha and Number mode (Note when the softkey says Alpha you are in number mode and when it says Number you are in alpha mode). Enter the phone's IP address, using number mode to enter the numbers and switching to alpha mode and using the * key to enter the dots in the IP address. When the number is entered correctly, press the Accept softkey.

Press the down arrow key to highlight the Subnet Mask item and if it is not correct, change this item in the same way that the IP address was entered.

Press the down arrow key to highlight the TFTP server item and if it is not correct, enter the address of the FacetPhone server.

Press the down arrow key to highlight the Default Router 1 item. Enter the IP address of the gateway that is used to route packets from the phone to other subnets on the LAN.

Press the down arrow key to highlight the DHCP Enabled item. If it is enabled, press the No softkey to disable it.

Press the Save softkey to save your changes to the Network Configuration settings.

Next, select MGCP Configuration on the main menu.

Select MGCP Gateway Controller and press the Edit softkey.

Enter the IP address of the FacetPhone system using the same procedure you used for entering other IP addresses.

Press the Save softkey to same the changes to the MGCP Configuration.

To relock the configuration, arrow down to Lock Config on the main menu and press the Select softkey.

Press the Back softkey to leave the configuration screen.

The display on the phone will continue to say "Configuring Phone" until the phone is defined in the FacetPhone system.
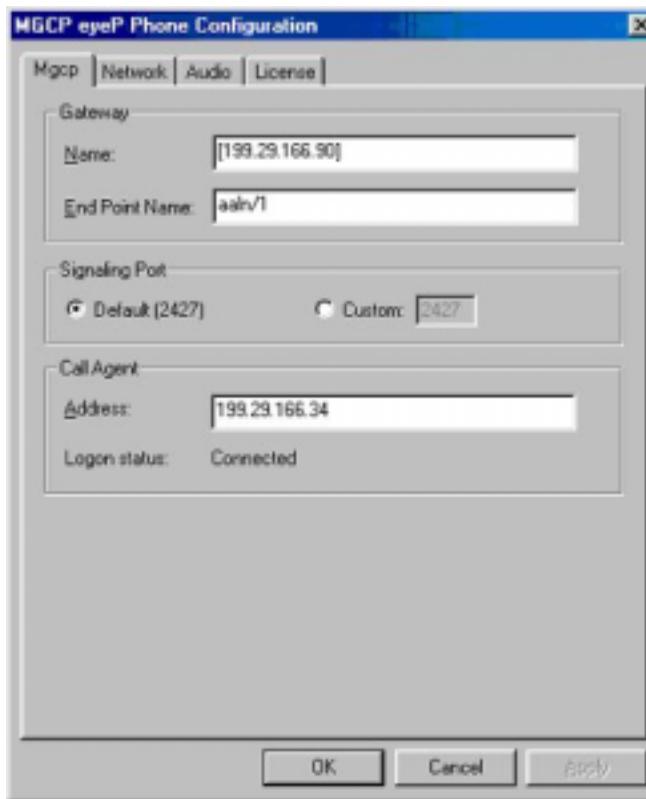
eyeP Media Soft Phone Setup

Install and run the eyeP Media software on your PC. Click on the menu button on the top left of the display:

On the menu that is presented, choose the "Configure" option.

This will bring up the configuration screen:

On the MGCP tab, enter the gateway name as the IP address of the PC within square brackets. For example:

[192.168.1.100]

Enter the endpoint name as:

aaln/1

Use the default signaling port of 2427.

Enter the IP address of the FacetPhone server as the Call Agent Address.

Next click on the Network tab:

The NAT/Firewall option is not supported.

The modem connection type may not give acceptable results depending on your modem connection.

The Audio Codec should be set to Automatic.

The RTP Ports should be set to Automatic.

Next click on the Audio tab:

This is where you select the audio devices on your PC that you will use to talk to and hear the other party.

If you check the "Use Voice Activity Detector" box it will cause the soft phone to only send voice packets when it detects you talking.

Finally, click on the License tab to enter your license information:

All other eyeP Media soft phone configuration will be done from the FacetPhone administrative interface as described in the "Defining IP Phones" section below.

Basic system configuration in the sys.local.cfg File

The most basic system configuration settings are kept in the files:

> /usr/facetphone/config/sys.cfg
> /usr/facetphone/config/sys.local.cfg
> /usr/facetphone/config/sys.branch.cfg

The sys.cfg file contains the default settings for the current software release. It is overwritten each time you upgrade your FacetPhone software.

The sys.local.cfg file is where you specify any settings that are unique for your site and different from the default values in sys.cfg. Any setting in sys.local.cfg overrides its corresponding setting in sys.cfg except for lists of things such user status definitions which are additive. Any user status definitions in the sys.local.cfg file will add to the ones already specified in sys.cfg. Any user status definitions in sys.local.cfg that have the same name as a definition in sys.cfg will be redefined as specified in the sys.local.cfg file.

The sys.branch.cfg file is used when there are multiple branches and one or more branches has a backup server. The sys.branch.cfg file contains those few entries that are unique for each branch's server.

The following settings must be customized for your site by putting an entry in your sys.local.cfg or sys.branch.cfg file. On a new system, the sys.local.cfg file will be created with some of these entries based on your answers to the questions asked by the installation procedure. Even if you are using the default value, it is best to put all of these entries in your sys.local.cfg file in case the defaults in sys.cfg change in the future. The sys.cfg file contains comments and the default values for each setting.

<htdocs_dir=/var/www/html>

This is the location of the HTML pages for your FacetPhone computer's web server and will be initially setup with the information you supplied at installation time

<tftp_dir=/tftpboot>

This is the location of the TFTP server's home directory on the FacetPhone server. This setting is currently only needed if Cisco ATA gateways are being used.

<default_group=Company>

Change "Company" to the name of your company or organization. FacetPhone creates a default group that includes all users and is the default operator group for all phones and lines. This setting causes the system to display your desired name for the default group.

<ca_addr=xxx.xxx.xxx.xxx>

The call agent address is the IP address that the gateways will use to address the FacetPhone server (the "call agent"). Enter the IP address of your FacetPhone server. If it has multiple addresses, enter the one that the gateways will use to communicate with the server. If your system has backup servers, then put this address in the sys.branch.cfg file.

<mail_server=xxx.xxx.xxx.xxx:25>

In order for FacetPhone to be able to forward voice mail to an email account or send voice mail notification to an email account, it must know the IP address and port of the SMTP server on your network. Enter that value here. You may enter a resolvable DNS name rather than an IP address. You should include the port number in either case. The default SMTP port is 25. Only change this if you know that your SMTP server is listening on a different port. If your system has backup servers, then put this address in the sys.branch.cfg file.

<user_low=100><user_high=199>
<phone_low=100><phone_high=199>

The default numbering plan allows for user and station extensions from 100 to 199. If your site is using a different numbering plan, then supply the correct values for these ranges. The user range and the phone range should be the same.

<speed_dial_low=20><speed_dial_high=59>

The default numbering plan reserves 40 two digit speed dial numbers from 20 through 59. You can reserve more or fewer numbers within any range you want as long as no speed dial number begins with the same digit that begins a user extension or station number. Speed dial numbers also cannot begin with the digits being used to access outside lines (8 and 9 by default). All speed dial numbers must be the same number of digits.

<line_low=1><line_high=50>

The range of numbers for telephone lines is specified in this line.

<max_group=30><max_pub_ustat=20>
<max_gw=10><max_call=100><max_text_chat=200>

These are the settings for the maximum number of groups, public user status definitions, gateways, concurrent calls and instant messaging sessions. You should size these appropriately for your site allowing some room for growth.

<max_nettapi_clients=100>

This setting is only required if you will be using software that integrates with FacetPhone using the NetTAPI protocol.

<max_branches=20>
<this_branch=0>

These settings are only needed if the system has multiple branches. The default value of max_branches is 20, so you only need to change this value if you have more than 20 branches. You should put the <this_branch> setting in the sys.branch.cfg file so that it is not duplicated on backup servers.

<outside_call_digit_map=(0T|[2-9]xxxxxxxxx|1xxxxxxxxxx|011x.T|911|1411|511|#)>

The outside calling digit map may need modification to match the dialing rules from your site. The rules for creating a digit map are:

> A digit map is a list of conditions to be matched separated by "|" characters and completely enclosed in parentheses.

> The digits "0" through "9", "*" and "#" in a match string means that the exact digit must be matched in that location.

> A range of numbers in brackets means that any number in the range will match in that location.

> An "x" means that any digit matches in that location.

> A "." means that a variable number of digits may occur in this location.

A "T" means that if a timeout period occurs with no digits having been entered, the digits entered so far will be reported as the dial string.

The default digit map is for an area where all local calls are done with 10 digit dialing as is common in many metropolitan areas now. However, the sys.cfg file includes settings that are commented out for locations with 7 digit dialing and locations for both 7 and 10 digit dialing:

```
# outside digit map for 10 digit local dialing
<outside_call_digit_map=(0T|[2-9]xxxxxxxxx|1xxxxxxxxxx|011x.T|911|1411|511|#)>
# outside digit map for 7 digit local dialing
#<outside_call_digit_map=(0T|[2-9]xxxxxx|1xxxxxxxxxx|011x.T|911|1411|511|#)>
# outside digit map for both 7 and 10 digit local dialing
#<outside_call_digit_map=(0T|[2-9]xxxxxx.T|1xxxxxxxxxx|011x.T|911|1411|511|#)>
```

Copy the appropriate digit map for your location in your sys.local.cfg file. If you will have your lines divided up into more than one trunk group, then you will provide the digit map for each trunk group in the trunk_groups.cfg file explained later.

If you changed the user ID range (user_low and user_high) such that it no longer includes IDs 198 and 199, then you must also edit the userid.cfg file and change the ID for the default administrative and FacetCorp support users. You may also want to change these default IDs if you plan to assign 198 or 199 to real users:

```
<uid=198><uname=admin>
<uid=199><uname=fc_support>
```

Change the uid values to IDs that are valid and available in your new user ID range.

After saving your changes, restart FacetPhone with the command:

    facetphone restart

You can make changes to your sys.local.cfg, sys.branch.cfg, sys_speed_dial.cfg and ivr.local.cfg files while the FacetPhone servers are running, and signal FacetPhone to reread these file. This is done with the command:

    facetphone reload

When FacetPhone rereads the sys.cfg, sys.local.cfg and sys.branch.cfg files while it is running, it ignores some fundamental system maximum settings. Specifically, the following settings are ignored when rereading these files:

```
<max_gw=n>
<phone_low=nnn>
<phone_high=nnn>
<speed_dial_low=nn>
<speed_dial_high=nn>
<line_low=nnn>
<line_high=nnn>
<user_low=nnn>
<user_high=nnn>
<max_group=nn>
<max_pub_ustat=nn>
<max_call=nn>
<max_parties_per_call=nn>
<max_text_chat=nn>
<max_nettapi_clients=nn>
<nbr_trunk_group_digits=n>
<allow_monitoring=y>
<ca_addr=xxx.xxx.xxx.xxx>
<ca_mgcp_port=nnnn>
<run_pbx=y>
```

There are quite a few other settings that can be made in the sys.cfg files, but this covers the ones most often changed during installation. Please contact FacetCorp support before changing any other settings in the sys.cfg files.

Defining the Branches in the branch.cfg File

If the FacetPhone system will have multiple branches, then these must be defined in the branch.cfg file. A branch is a different physical location that is connected to headquarters over the company WAN. For example, a company with offices in multiple cites would define each office as a branch in the FacetPhone system. Similarly, a telecommuter's location would also be defined as a branch.

Each branch is given a branch number with the headquarters branch being branch 0. Each branch entry looks like this:

> <branch=0><name=Headquarters><server_type=active>
> <branch=0><server_addr=199.29.166.34><ctrl_branch=0><tz=CST6CDT>

The items can be in any order on as many lines as desired as long as each line begins with the <branch=> specification. The meaning of each item is:

> branch=

This item species the branch number. The headquarters branch should be branch 0.

> name=

This is a description of the branch that would probably indicate the branch's location.

> server_type=

This specifies whether there is a server at the branch and if so whether it is an active or backup server. The values for this setting can be "active", "backup_local", "backup_all" or "none". If the server at the branch is the main server for the entire system, its server_type will be "active". If it is serving as a backup for its local branch only then it will be "backup_local" and if it is serving as a backup for the whole system, it will be "backup_all". If there is no server at the branch, the server_type is "none". When the facetphone command is used to have a server take over or give up control, this setting will automatically be changed in the branch.cfg file.

> server_addr=

This specifies the address of the FacetPhone server at the branch. If there is no server at the branch, then this value is left blank.

ctrl_branch=

This setting specifies which branch has the server that is controlling the branch being defined. Under normal operation all of the branches will be controlled by branch 0, the headquarters branch. When the facetphone command is used on a FacetPhone server to take over or give up control of one or more branches, this setting will be set automatically.

tz=

This specifies the time zone for the branch in standard UNIX/Linux notation where the standard time zone abbreviation is followed by the number of hours west of GMT followed by the daylight savings time zone abbreviation. For example Central time in the U.S. is specified as CST6CDT.

Defining Trunk Groups in the trunk_groups.cfg File

The trunk_groups.cfg file contains the definition for each trunk group. Lines are organized into trunk groups for the purpose of choosing which lines to use for outgoing calls and to specify the destination for DID (Direct Inward Dial) numbers that are incoming on a group of lines.

Each trunk group is defined by the following entries in the trunk_groups.cfg file.

<trunk_group=0><desc=Main lines>

Each entry in the trunk_groups.cfg file begins with <trunk_group=*n*> where *n* is the trunk group number. The "desc" value is just a description of the trunk group.

<trunk_group=0><outside_call_digit_map=(0T|[2-9]xxxxxxxxx|1xxxxxxxxxx|011x.T|911|1411|511|#)>

The "outside_call_digit_map" is the digit map as described in sys.local.cfg. If no digit map is specified here, then the one in sys.local.cfg will be used for the trunk group. If an outside digit map is specified for a trunk group, then the digit map specified here overrides the one specified in sys.local.cfg when this trunk group is being used for an outgoing call.

<trunk_group=0><did=6910><type=user><dest=110>

Each DID number provided by a T1 is specifed by a "did" entry. The value of the "did" entry is the digit string presented by the phone company. Sometimes this is the entire number and sometimes it is part of it, such as the last 4 digits dialed. DID numbers can be routed to a user, a station or a user group. When routed to a user, the type is "user" and the dest (destination) is the user extension that should be called. A call dialed to the user's DID will ring the user if he is available for a call and otherwise the call will go to his voice mail following the user's voice mail configuration. If a DID number is routed to a station, then the type is "station" and the dest value is the station number. A call dialed to the station's DID will ring the station with no voice mail backup. If a DID number is routed to a user group, the type is "group" and the dest value is the name of the group. A call dialed to the group's DID number will follow the auto-attendant rules currently in effect for the group.

<trunk_group=0><outgoing_cid_number=9729859901>

On a trunk group being used for a T1, the caller ID number to be used on outgoing calls can be specifed as the value of the "outgoing_cid_number" element.

Editing the sys_speed_dial.cfg File

The final configuration outside of the FacetPhone administrative interface is the system-wide speed dial numbers. These are kept in the file:

/usr/facetphone/config/sys_speed_dial.cfg

Each entry in this file looks like:

<speed_dial=20><number=9,9728441111><desc=Time & Temperature>

The "speed_dial" number is what the caller presses to dial the outside "number" specified in the next field. The outside number must include the leading 9 or 8 plus trunk group that will be used to select the outgoing line for the call. The comma used in this example is not necessary. It is only indicating that it is the leading 9 for the default trunk group rather than part of the outside number. The "desc" value is a description of 100 characters or less. The description cannot contain the angle brackets that are used to delineate each field of the entry.

After updating the sys_speed_dial.cfg file, if the FacetPhone servers are running you can signal the fp_db process to reread this file as described above in the sys.local.cfg section.

Windows PC Installation

The remainder of the configuration will be done with the FacetPhone User Interface. The installation of the FacetPhone client software on a Windows PC is described in the FacetPhone Users Guide.

Running the FacetPhone User Interface

If your desktop PC is running Windows, the FacetPhone installation procedure will have put an icon on your desktop that is called "FacetPhone". Double click this icon to start the user interface.

When the user interface starts for the first time it will present the Connection Configuration screen:
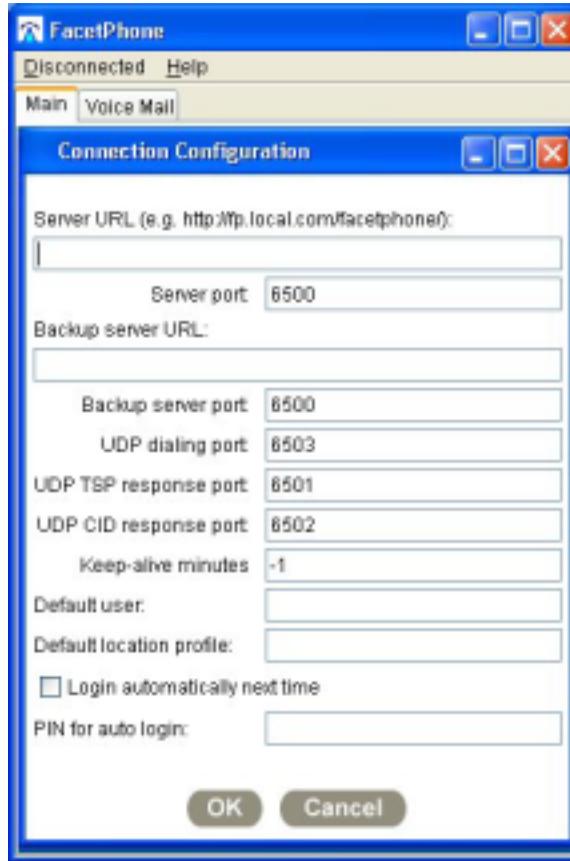
Enter the URL to your FacetPhone server followed by "/facetphone". You may use its IP address if it is not named in a DNS. For example:

> htttp://199.29.166.34/facetphone

Replace the example IP address in the URL above with your FacetPhone server's IP address or DNS name.

You can keep the default values for the port settings on this screen.

If the same user will always be using this PC to login to FacetPhone, you may enter that user's login name in the default user setting and this name will be automatically filled in on the login screen. This can be overridden at login time.

You can also enter the name of the location profile that you will normally want to use when at this PC and it will be filled in for you. You can also override this setting at login time.

Click OK to the connection configuration screen once all the settings are entered.

The login screen will be presented:

When initially configuring the system, enter "admin" for the user name and leave the pin blank. Click on the Login button.

Next the Location Selection screen will be presented:
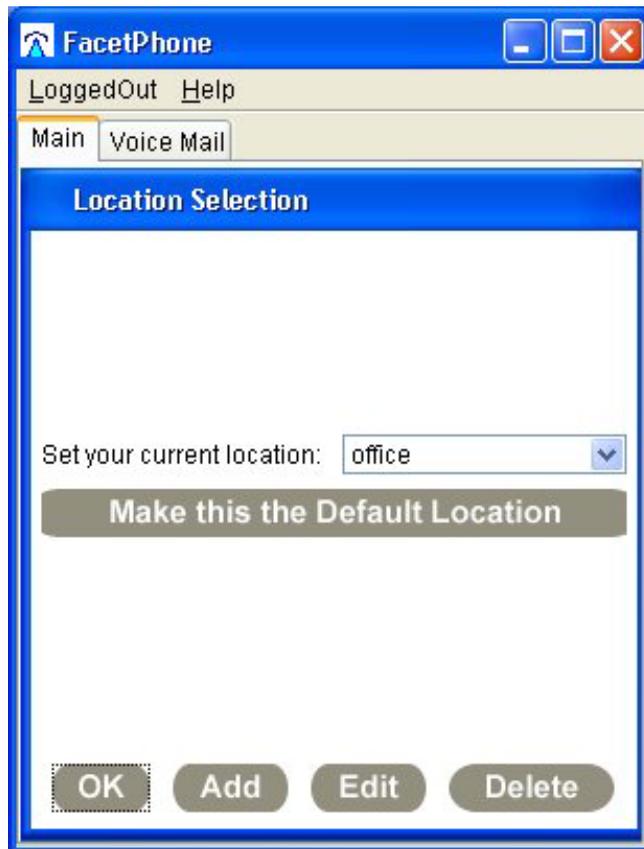
The "office" location should be selected. Click OK to proceed with the login.

The main user interface screen will be presented:
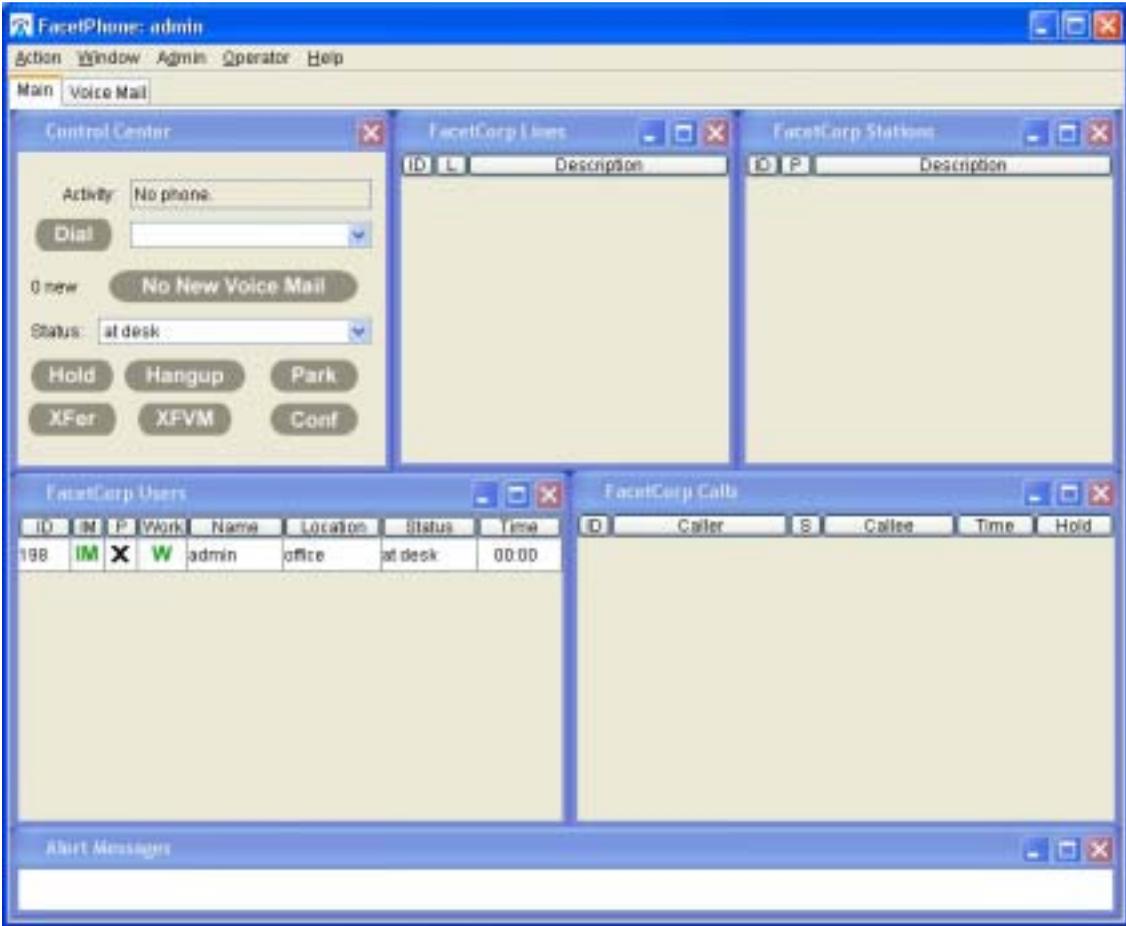
Defining Gateways

On the user interface menu bar, choose the menu selection:

Admin / Change System Configuration
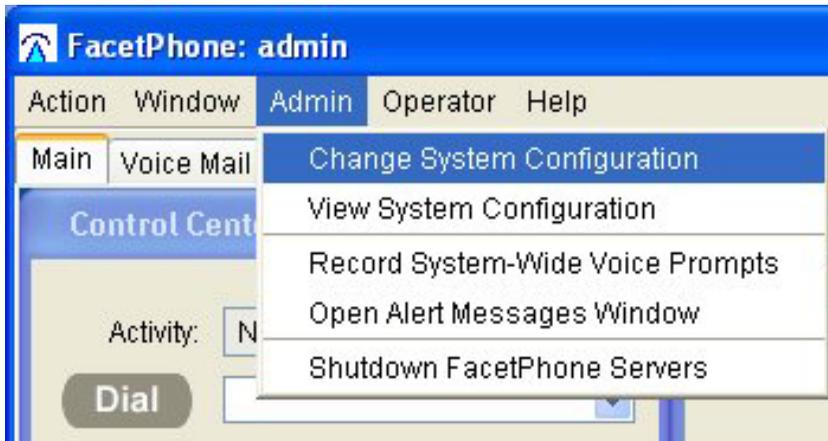
The administration window will be opened:

Only one administrator may make changes to the system at a time. If someone else has the administrator locked, a message will be displayed with the user name of the person who has it in use.

On the administration window's menu, select

> Add / Add a Gateway

This will present the gateway definition dialog box:

Each gateway is given a numeric ID. You can enter the number in the "Gateway ID" field or check the box to have the next available ID used. Gateway IDs begin with 0, but you may find it less confusing to number them beginning with 1.

Name: This is for information only. The system will generate the name of the gateway.

Gateway description: Enter a description of the gateway. We suggest that the description include the gateway manufacturer and the number and type of ports it has, for example:

Voipack 2x2

IP address of the gateway: Enter the IP address that the gateway was configured with.

MAC address of the gateway: The MAC address is usually printed on the device somewhere. If you cannot find it, this field can be left blank for most types of gateways.

Default route: The default route is the IP address of the network gateway that can route packets if the FacetPhone server and/or other FacetPhone VoIP gateways are not on this subnet. If the gateway is on the same subnet with the server and all other gateways, this field may be left blank.

Manufacturer: Choose the gateway manufacturer from the list.

Number of FXS ports: Enter the number of FXS ports the gateway has. The FXS ports are the ones where telephone instruments will be plugged in. Voipack gateways which can have their ports configured for FXS or FXO should always be configured to have any FXS ports configured to be the first ports before any FXO ports.

Number FXO ports or T1 Channels: Enter the number of FXO ports the gateway has if the gateway will have analog lines from the phone company plugged into it. If it is a T1 gateway, enter the total number of channels on the T1(s) that will be plugged into it .

Branch where gateway located: Select the branch where the gateway is located from the list.

Gateway can generate DTMF tones on the line: Checking this box means that the gateway can be told to play an outgoing DTMF digit rather than having to pass the audio representation of the digit from the telephone instrument connected to a call. This setting is relevant to FXO ports or T1 gateway channels. It is usually checked.

Gateway should only be sent VMWI when it changes: This is only set for gateways that do not properly handle multiple redundant VMWI commands. This setting is only relevant to FXS ports. You can usually leave this box unchecked.

Gateway supports volume changes in call: This can be set for FXS gateways that support commands to change the volume while a call is connected. If this is checked, then a user whose phone is on this gateway will be presented with a volume slider bar on their FacetPhone user interface. This changes the receiver (handset, headset, speaker) volume not the microphone volume.

Digits detected in audio: This is used to specify that an FXO or T1 gateway will not detect digits and the IVR process should detect them from the audio stream. In the case of an FXS gateway, it means the gateway will not notify of digits pressed on the phone while in a connection and the digits will be sent in the audio stream.

RFC 2833 digit detection: This is used to specify that an FXO or T1 gateway will detect the digits and send them in as special RTP packets that the IVR process recognizes. In the case of an FXS gateway, it means that digits pressed on the phone while in a connection will be transmitted by the gateway as RFC 2833 packets in the RTP stream.

The "SIP trunking service settings" are only relevant if this is a virtual gateway to setup a connection to a SIP trunking service (like Vonage, but a business class service that does not require the service provider's own physical gateway).

Gateway ID to consolidate lines: This is used when multiple SIP trunking accounts that require their own SIP trunking settings have their lines consolidated onto a single virtual gateway.

URI Domain: This is the domain name that will be used in an INVITE message to the SIP trunking service.

Registrar: This is the domain name used in registration messages to the SIP trunking service.

Number: This is the phone number used in the first part of the number@domain address used in To, From and Contact headers in messages to the SIP trunking service.

ITSP Domain: This is the domain name used in the second part of the number@domain address used in To, From and Contact headers in messages to the SIP trunking service.

Username: This is the user name used in the Authorization header to the SIP trunking service.

Password: This is the password used in the Authorization header to the SIP trunking service.

Expires: This is the expiration period in seconds used as the "expires=" value in the Contact header to the SIP trunking service.

NAT Address: The NAT Address specifies the IP address that the voice packets sent to this SIP trunk will appear to come from when not using a SIP proxy server.

Use E164 phone number format

Click OK on the gateway dialog box.

The gateway name is automatically created from the other information you enter. When you save your changes to the gateway configuration, if it is a Voipack gateway, it will automatically be programmed to have its new name. Please wait until this operation is complete before continuing.

On the administrator window menu, choose

       Configure / Configure Gateways

This will present a list of the gateways defined. The new gateway should be in the list:

To make a change to an existing gateway, you can click on a gateway in this list and choose the Edit function from the menu presented.

If the gateway is a Voipack gateway, click on the gateway in the list and click on the "SEND settings" item from the menu that is presented.

If you have more than one gateway, repeat this procedure until all the gateways have been defined.

You can close the gateway list window by clicking on the close [X] button on its title bar.

Defining Groups

Configuring groups is done in two parts when initally setting up a system. At this point we will define the groups and then after the users are all added to the system, the groups will be edited again to assign group memberships.

The default group is predefined during the FacetPhone installation. This group will appear in the group list with the name that it was given in the sys.local.cfg setting. In this example, we have provided "FacetCorp" as the default group name:

<default_group=FacetCorp>

We will configure the default group first and then add new groups as needed.

On the administration window's menu, select

Configure / Configure Groups

This will present the "Group Configuration Selection" window:
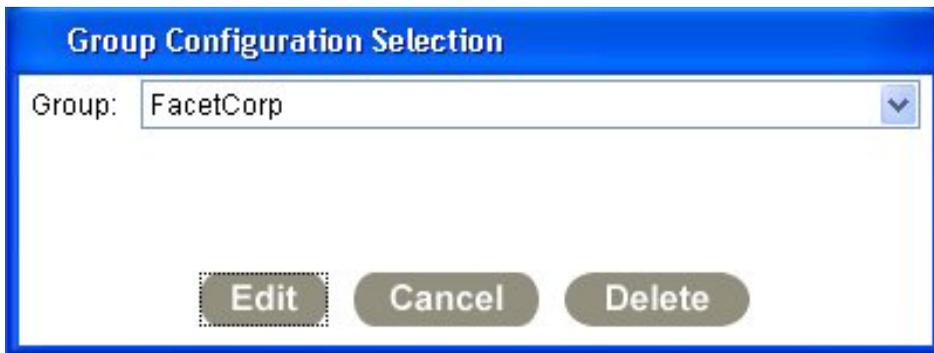
Click the Edit button to display the configuration screen for the default group:

Group FacetCorp

Group name: FacetCorp

☐ All users   **Members**   **Operator Stations**

☑ Anyone can answer any ringing call

☑ Managers can monitor and barge in

Ring for new incoming calls `-1`

Ring for transferred calls `0`

Parked call ring back seconds `0`

Home branch: Headquarters ▾

Auto-attendant settings

☐ Auto-attendant on

ON script name: company_open_menu ▾

OFF script name: company_overflow_menu ▾

Rings before answer when OFF: `6`

ON HOLD script name: music_on_hold ▾

Automatic call distribution: Off ▾

Group display privacy settings

User list privacy level: Members can watch ▾

Call list privacy level: Members can watch ▾

Mailbox privacy level: Members can watch ▾

Station list privacy level: Members can watch ▾

Parked/queued calls privacy level: Members can watch ▾

Lines list privacy level: Members can watch ▾

Voice Mail Handling

OK   **Cancel**

You will not enter group members or operator stations yet, since no users or phones are defined. At this point we just want to define the groups so that they are available to specify as operator groups for lines and phones as we create those. You should also enter whatever other information you have about the group at this time.

Anyone can answer any ringing call: If this box is checked then any user in a group associated with the call can pickup any call ringing in their branch by dialing #0 on their phone or by clicking on the call in the user interface call display and selecting "Pickup" on the menu presented.

Managers can monitor and barge in: This box must be checked in order for managers in this group to monitor members of the group who are not managers. In addition to this setting for each group, monitoring and barge-in require that the sys.local.cfg file have the entry:

<allow_monitoring=y>

A change in the allow_monitoring setting requires a restart of FacetPhone in order to take effect.

Distinctive ring: This is the ring used when a call is ringing to the operators for the group. A value of 0 is a single ring, 1 is a double ring and 2 is a triple ring. Values other than this are unique for each gateway or IP phone type. If you have incoming calls to different groups, you may want to give each group a different ring type so that an operator taking calls for multiple groups can distinguish between the groups being called based on the ring. Note that the default ring type for a station-to-station call is 1 which is two short rings during each ring cycle. If you want to change the station-to-station ring type, put the line

      <inside_ring_type=*n*>

in your sys.local.cfg file where *n* is the ring type for the station-to-station calls. A distinctive ring value of -1 for a group or for the inside ring type will cause the phone to use the ring type that the phone has been configured to use. For example, the Polycom IP phones allow the selection of a ring type from the phone's local menu.

Ring for transferred calls: This setting allows the specification of a different ring type for calls that have already been answered by someone and are now ringing again because the call is being transferred, ringing back from being parked, etc.

Parked call ring back seconds: This specifies the number of seconds that the call is parked before ringing back to the operator group. Note that this is used in conjunction with music on hold and the call will only be checked for ring back after a song has finished playing.

Home branch: This setting determines the time zone used to interpret the time of voice mails left for the group.

The Auto-attendant settings define the operation of the auto-attendant for any lines and stations that are controlled by this operator group. In addition, it has the setting to specify the IVR script that will be run when a caller to this group is put on hold or parked.

Auto-attendant On: If this box is checked, then the auto-attendant will be invoked immediately for an incoming call.

ON Script name: This specifies the auto-attendant script you want when the auto-attendant is on. Usually you will create scheduled events for turning the auto-attendant on and off and selecting the script. For example on business mornings you would have the auto-attendant turned off or set to run a company open script. At the end of business days, the auto-attendant would be turned on and a company closed script would be selected.

OFF Script name: This is the script that is played after some number of rings when the auto-attendant is off. This script is also played if all the operators are busy and do not have call waiting turned on.

Rings before answer: This is the number of rings to wait when the auto-attendant is off before invoking the auto-attendant "off" script.

ON HOLD Script name: This determines what IVR script, if any, will be run when a caller to this group is put on hold or parked. If <none> is selected, then no IVR script is run and callers on hold or parked will hear silence. The default scripts supplied for this purpose are "message_on_hold" and "music_on_hold". The message_on_hold script will play the prompt "on_hold_message" which can be recorded from the Operator menu. The music_on_hold script will randomly choose from among 10 songs to play. Ten royalty free songs are installed as defaults.

Automatic Call Distribution: This setting indicates whether incoming calls will be distributed to one operator at a time in round robin order, one at a time in a fixed order or all operators rung simultaneously when a new call comes in. Operators must have their status set to "oper on duty" in order to receive incoming calls. If automatic call distribution is "Off", then all the operators for the group who are available for a call will be rung at once. If it is set to "Round Robin", then the next available operator after the one previously rung will receive the new call. If it is set to "Fixed Order" then the ring

order specified on the Members page will used. The first person in the order will always be rung first if they are available. If they are not available, then the list will be searched in the same order each time to find the next available operator.

The privacy levels determine what user privileges are required to view certain displays in the user interface. You can use these settings to determine who can watch a list of users, calls, parked or queued calls, stations and lines. The settings are hierarchical where managers can watch anything operators are allowed to watch and operators can watch anything that members are allowed to watch. Operators and managers of the group are designated on the screen presented when you click the Members button as described below.

To setup the voice mail handling for the group, click on the Voice Mail Handling button on the group edit screen:

The default is to save voice mail in the group's voice mail folder only. This makes the voice mail accessible from the FacetPhone user interface, from a telephone on the system and from calling in on an outside line to the system.

To have the group voice mail sent as a .wav file attachment to an e-mail, check the "Voice mail to e-mail" box and enter the email address where voice mail messages should be sent. Multiple email addresses may be specified and must be separated by a semi-colon or comma.

If you want to have a notification message sent to an e-mail address when a new group voice mail is received, then check the "Notify to e-mail" box and enter the address where the voice mail notification e-mail should be sent. The notification email will include a subject line with the caller ID information, but will not include any email message body. This option does not send the actual voice mail message as a .wav attachment. Multiple email addresses may be specified and must be separated by a semi-colon or comma.

If you want the system to call a phone number or a list of numbers when a new group voice mail is received, then check the "Notify by phone" box and enter the phone number(s) to call including the leading 9 or 8 plus trunk group number. To call a list of numbers, separate the numbers with a comma. The system will continue to call the numbers one at a time until someone answers and presses *. The system will then prompt for a user extension and PIN. At that point the notified person will be in the user menu and can listen to and manage the new voice mail.

To have the system send an SMS text message to notify of a new group voice mail, check the box "Notify to SMS" and enter the SMS address where the notification should be sent, usually phonenumber@provider. The text message will contain the same information that an email notification contains. Multiple SMS addresses may be specified and must be separated by a semi-colon or comma.

If you have selected email delivery of the message, email notification or SMS notification, then enter a valid e-mail address in the "From address for e-mails".

Click on the OK button to save your changes and return to the main group configuration screen.

Finally, you will need to define the members, operators and managers for the group. If you have not yet defined the users or stations, you should do that first and then return here to specify the group membership and operator stations.

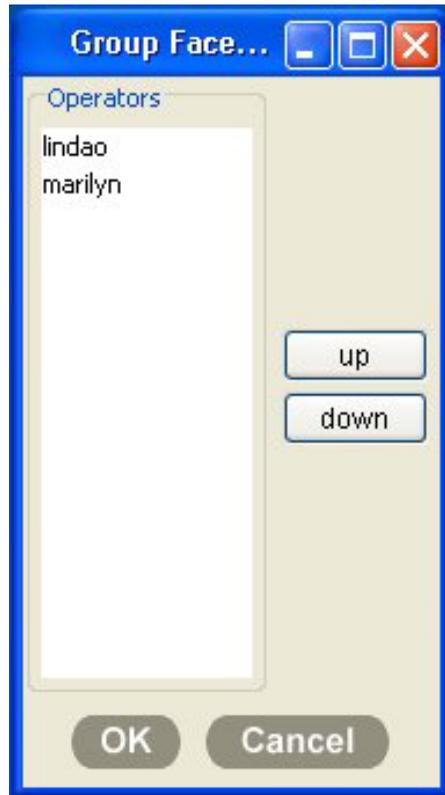Click on the Members button and the group membership dialog box will be displayed:
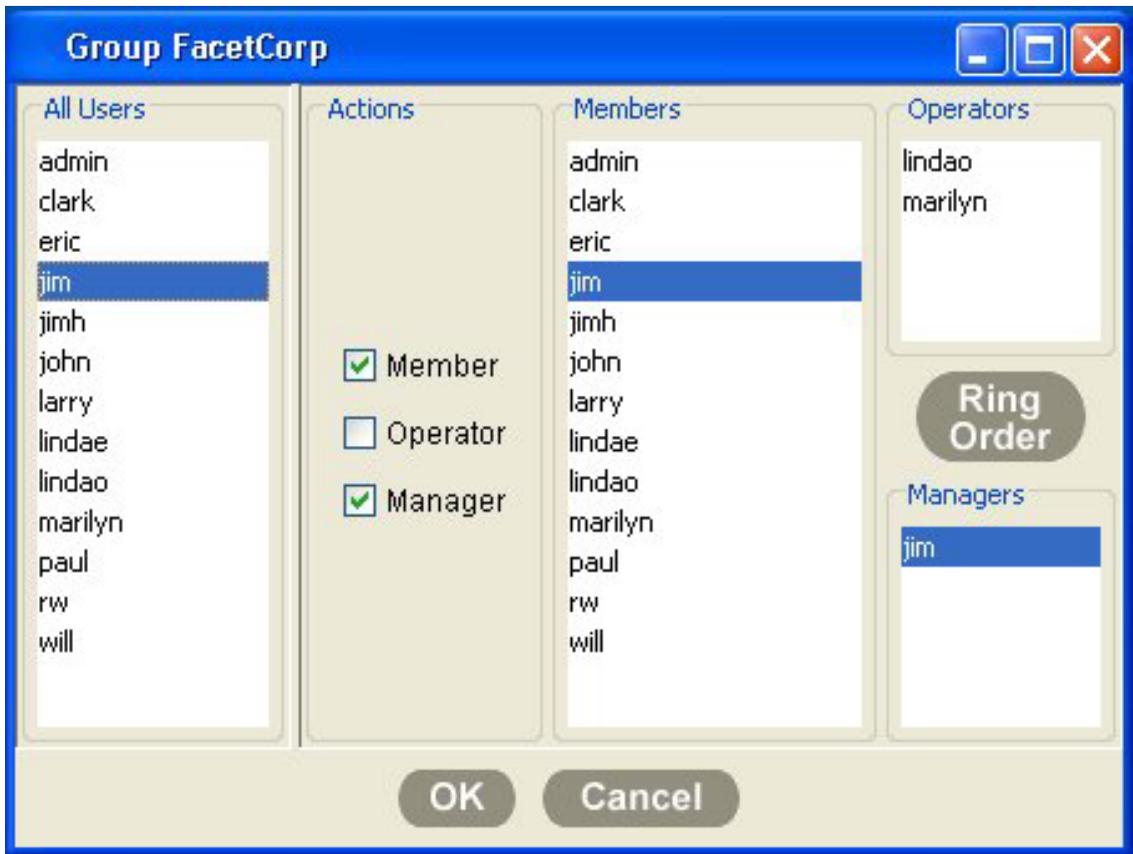
To associate a user with the group, click on the user's name in the "All Users" list and then check the appropriate boxes to make them a member, operator or manager. You will usually want to check the "User" box for operators and managers in addition to the "Operator" or "Manager" boxes. A person can be both an operator and a manager.

Operators will have incoming calls to the group directed to their phones wherever they are logged in. The operators must also have their status set to "Oper on duty" in order for incoming calls to ring to them. Operators also get the Operator menu items included on their user interface.

The Manager designation is used to allow monitoring and barge-in on calls in the group by the person identified as a manager for the group.

If the group is using "Fixed order" automatic call distribution, then you will want to click on the Ring Order button to arrange the order that the operators for the group will be rung:

To move an operator in the list, click on their name and then click on the up and down buttons to move them up and down in the list. When the ring order is the way you want it, click OK on this window and click OK on the membership window to return to the main group edit window.

Next, in the main group edit window click on the "Operator Stations" button. The operator stations display will be presented:
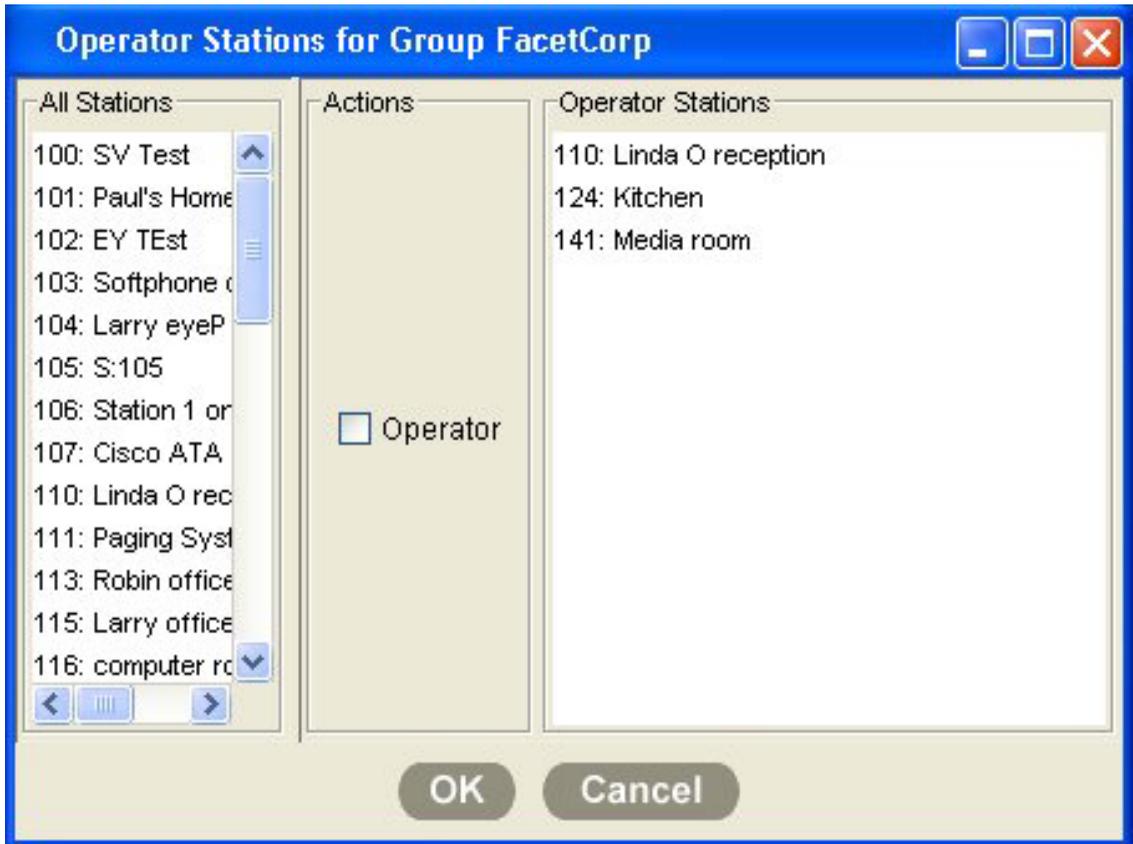
You can select a phone to ring on incoming calls to the group even if nobody is currently logged into that station. For example, here at FacetCorp, we not only have several people defined as operators, but we have several phones scattered around the office defined as operator phones (such as the kitchen and copy room). This allows incoming calls to be answered in a variety of stations around the office. After defining any operator stations you want, click OK to close this window.

Click OK on the main group edit window.

You can repeat this procedure for each group you wish to create.

Defining IP Phones

On the administration window's menu, select

> Add / Add an IP Phone

This will present the IP phone definition dialog box:

Station number: Enter the station number for the IP phone being defined or check the box to assign the IP phone the next available station number.

Manufacturer: Choose the IP Phone manufacturer and model from the list.

IP phone description: Enter a description for the phone. We suggest that you include information about the location of the phone.

Dynamic IP address: If this box is checked, then the phone will be assumed to have a dynamic IP address. This is currently only supported for soft phones. Any soft phone given a dynamic address must be configured to have its name set to "stationnnn" where nnn is the phone's station number on the FacetPhone system.

IP address of the IP phone: If the phone will have a fixed address, enter the IP address that the phone was configured with.

MAC address of the IP phone: You can enter the MAC address of the phone in the space provided. Currently this information is not necessary for any of the IP phones supported by FacetPhone.

Default route: Enter the default route that the IP phone should use to route packets if the server and/or other gateways are not on its subnet. If the phone is on the same subnet with the server and all other gateways, this field may be left blank.

NAT address: If necessary for an IP phone not on the LAN, enter the public IP address that packets sent to the IP phone will appear to come from.

Branch where IP phone located: Select the branch where the phone is located from the list.

Operator group: Choose the operator group that will control this IP phone. This determines which operators will be rung if a user dials "0" on this phone. It is also used to determine which calls can be picked up by the phone if the system is configured to restrict calls to the original called group.

Dialing restriction: Select a dialing restriction or <none> from the drop list. This will be the dialing restriction, if any, to be used on this phone when there is no user associated with the phone. If a user is associated with the phone, the user's dialing restriction will override the phone's dialing restriction.

Default outgoing trunk group(s): Choose the default outgoing trunk group(s) for the IP phone. This determines which outside lines will be used by when a user makes an outside call from this phone. You can enter a list of trunk group numbers separated by commas. The trunk groups in a list will be searched in order until an available line is found for the call.

Digits transmitted in audio: This is used to specify that the phone will not notify of digits pressed while in a connection and the DTMF will be sent in the audio stream.

Transmit digits as RFC 2833 packets: This is used to specify that the phone will send RFC 2833 packets in the RTP stream to indicate digits pressed while in a connection.

Route all audio through the server: If this option is in use, all packets sent to and from this phone will be routed through the FacetPhone server instead of being sent directly to and from the other endpoint in the conversation.

Route audio through the server for different branch: This option will cause the audio packets to and from this phone to be sent through the server only when the other endpoint is at a different branch.

Click OK to save the new IP phone definition. Repeat this procedure for each IP phone.

To see a list of all IP phones defined, select

> Configure / Configure IP Phones

on the administration window menu . This will present a list of all the IP phones you have defined:

You can click on an IP phone and choose Edit from the menu to make changes to it.

Defining Stations (Phones) Connected to a Gateway

On the administration window's menu, select

Add / Add a Station (to a Gateway)

This will present the station definition dialog box:
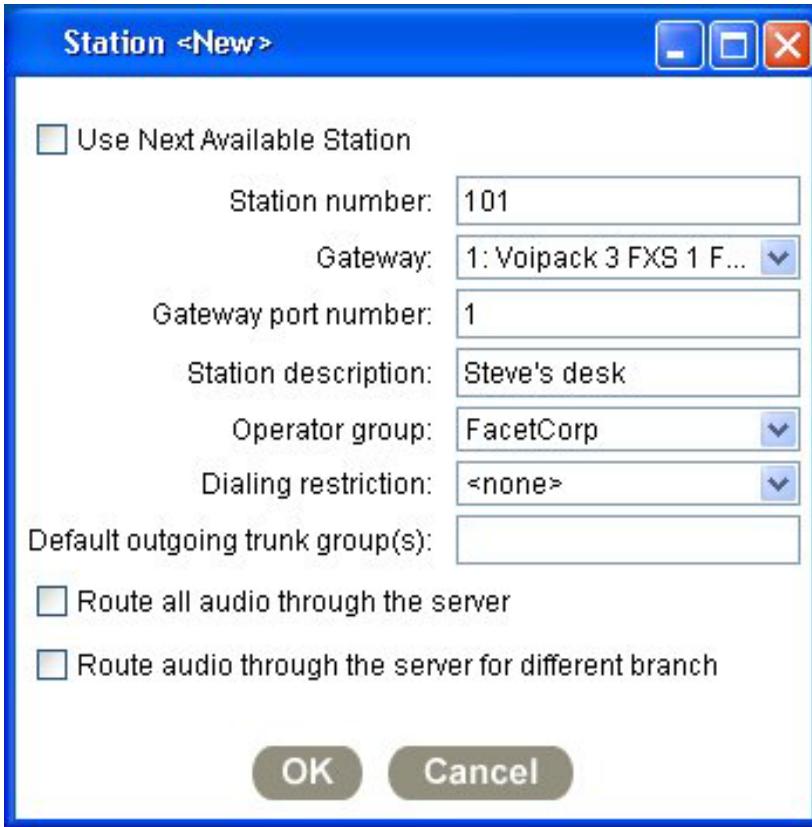
Station number: Enter the station number for the phone being defined or check the box for the station to be assigned the next available station number.

Gateway: Choose the gateway that this station will be connected to from the drop list of the gateways you have defined.

Gateway port number: Choose the port number that the station will be connected to. Ports are numbered beginning with 1 from left to right on the gateway. Note that stations must be connected to FXS ports only.

Station description: Give the station a description. We suggest that you include information about the location of the phone plus information about the gateway it is connected to. For example, "Warehouse GW1:1" meaning the phone in the warehouse connected to gateway number 1, port number 1.

Operator group: Choose the operator group that will control this station. This determines which operators will be rung if a user dials "0" on this phone. It is also used to determine which calls can be picked up by the phone if the system is configured to restrict calls to the original called group.

Dialing restriction: Select a dialing restriction or <none> from the drop list. This will be the dialing restriction, if any, to be used on this phone when there is no user associated with the phone. If a user is associated with the phone, the user's dialing restriction will override the phone's dialing restriction.

Default outgoing trunk group(s): Choose the default outgoing trunk group(s) for the station. This determines which outside lines will be used by when a user makes an outside call from this phone. You can enter a list of trunk group numbers separated by commas. The trunk groups in a list will be searched in order until an available line is found for the call.

Route all audio through the server: If this option is in use, all packets sent to and from this phone will be routed through the FacetPhone server instead of being sent directly to and from the other endpoint in the conversation.

Route audio through the server for different branch: This option will cause the audio packets to and from this phone to be sent through the server only when the other endpoint is at a different branch.

Click OK to save the new station definition. Repeat this procedure for each station.

To see a list of all stations defined, select

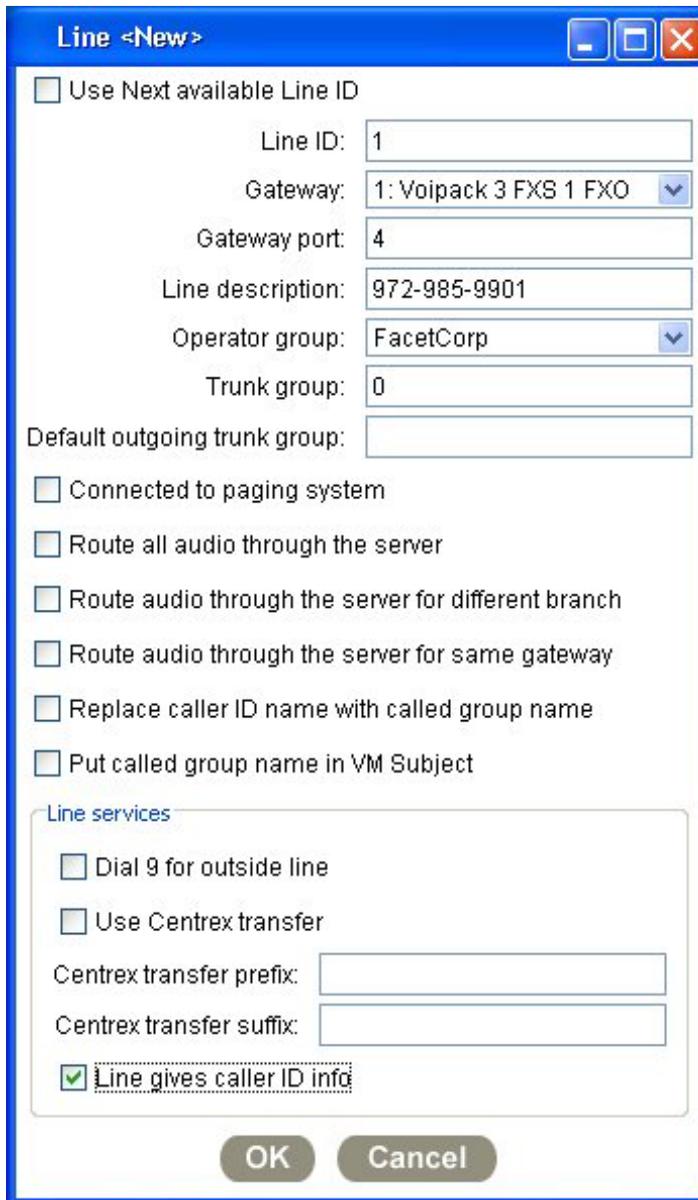 Configure / Configure Stations (on Gateways)

on the administration window menu . This will present a list of all the stations you have defined. You can click on a station and choose Edit from the menu to make changes to it.

Defining Lines

On the administration window's menu, select

> Add / Add a Line

This will present the line definition dialog box:

Line ID: Enter the line number for the line being defined or check the box for the line to be assigned the next available line number.

Gateway: Choose the gateway that this line will be connected to from the drop list of the gateways you have defined.

Gateway port: Choose the port number that the line will be connected to. Ports are numbered beginning with 1 from left to right. Note that lines must be connected to FXO ports only.

Line description: Give the line a description. We suggest that you include the line's phone number, the trunk group it is assigned to and information about the gateway it is connected to. For example, "972-985-9901 TG0 GW1:3" would show the line's phone number, that it is in trunk group 0 and that it is connected to gateway 1, port 3.

Operator group: Choose the operator group that will control this line. This determines which group's auto-attendants and operators will be used when a call comes in on this line. Note that if this is a digital line with DIDs, then a call coming in on the line may be directed to a different group if the DID is defined to do so in the trunk_groups.cfg file.

Trunk group: Choose the trunk group for the line. Trunk groups organize lines into groups for selecting a line for an outgoing call. Trunk groups also specify DID numbers and their routing on incoming calls.

Default outgoing trunk group: The default outgoing trunk group for the line is used when an incoming call on the line is forwarded back out on another line. The default outgoing trunk group will be used to select the outgoing line. You can enter a list of trunk group numbers separated by commas. The trunk groups in a list will be searched in order until an available line is found for the call. Note that if the line is configured to do a Centrex transfer, no additional line will be needed and this setting is not used.

Connected to paging system: If this FXO port will be connected to a paging system instead of an analog line from the phone company, then check the box. This setting causes a flash from the paging system to be treated like a hangup.

Route all audio through the server: If this option is in use, all packets sent to and from this line will be routed through the FacetPhone server instead of being sent directly to and from the other endpoint in the conversation.

Route audio through the server for different branch: This option will cause the audio packets to and from this line to be sent through the server only when the other endpoint is at a different branch.

Route audio through the server for same gateway: This option will cause the audio packets to and from this line to be sent through the server only when the other endpoint is on the same gateway.

Dial 9 for outside line: If the line has service that requires that a phone connected directly to the line must dial 9 before getting an outside dial tone, check the box for that feature.

Use Centrex transfer: If the line has Centrex transfer that allows you to hook flash to put an outside party on hold, dial another outside number, and then hang up in order to transfer the party to the new outside number, check the box for that feature. If the feature requires digits to be dialed before the hook flash, enter those digits as the "Centrex transfer prefix". If the feature requires digits to be dialed after the hook flash, enther those digits in the "Centrex transfer suffix".

Line gives Caller ID info: If the line has caller ID service, check the box for that feature.

Replace caller ID name with called group name: If you want to replace the incoming caller ID name on this line with the name of the group that was called with a DID number, then check the box for that feature. The caller ID number will be displayed as usual.

Put called group name in VM subject: If you want to add the called group name to the subject line of any group voice mails left on this line, then check the box for that feature.

Volume (-13 to 13): The volume setting may be used to change the volume of what is heard by the outside party on the line. This feature is only available for lines connected to a Voipack gateway.

Click OK to save the new line definition. Repeat this procedure for each line.

To see a list of all lines defined, select

> Configure / Configure Lines

on the administration window menu . This will present a list of all the lines you have defined. You can click on a line and choose Edit from the menu to make changes to it.

Defining Users

There are many details defined for each user including the complete description of all the user interface displays and their location in the main user interface window. Therefore, instead of defining a user from scratch, you always begin by creating a new user from an existing user or user "model". On the administration window's menu choose Configure / Configure Users. The admin user which was installed by default should be in the list of users. In addition there should be two user models "operator" and "user". A model is identified in the list by the fact that it has asterisks in its ID column. You can create a new user like any existing user or user model by clicking on its entry in the list and choosing "Make a NEW user like …" from the menu that is presented. Users will always be able to reconfigure their own settings except for their privileges which must always be set by an administrator. For this example, create an operator's account by clicking on the operator model and choosing "Make a NEW user like operator" from the menu:
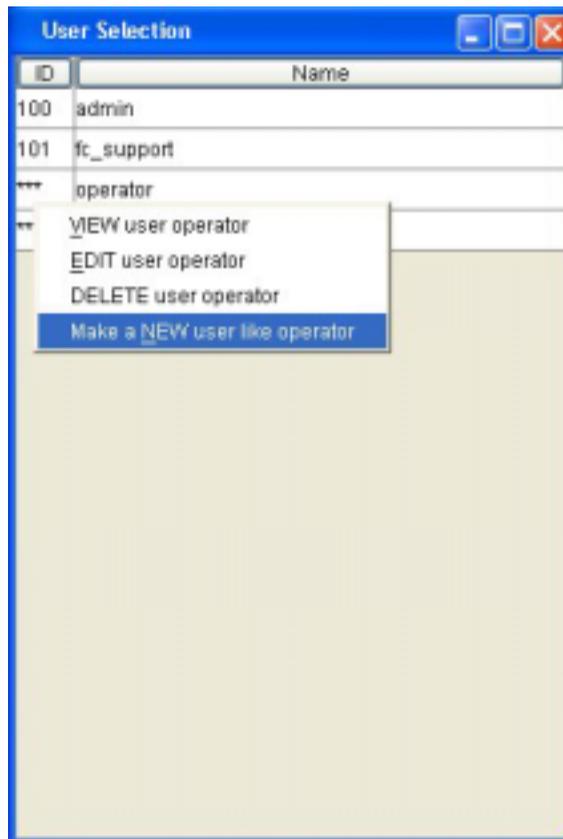
A new user definition screen will be presented where you will enter the user's extension number and login name:
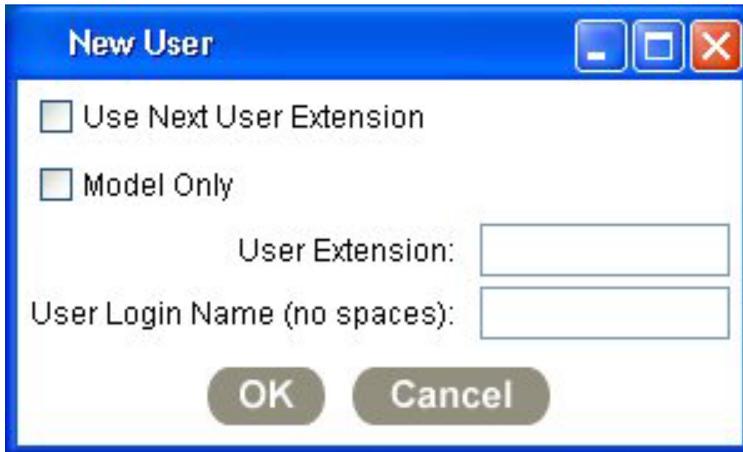
User Extension: If you want the system to use the next available user extension number, check the box for that. Otherwise, enter the user's extension number.

Model Only: If the new user being created will be a user model and not an actual user, check the box for that.

User Login Name (no spaces): Enter the user's login name that will be used when the user logs into the FacetPhone user interface. If the user already has a user name on other computers, you probably want to use that same user name here. The user name can have up to 20 characters, cannot contain spaces, tabs or control characters and is case sensitive.

Click OK on this dialog box and then the complete user definition dialog box will be presented with the user's extension number and login name already filled in as you entered them:

User jim

User extension: 144

User name: jim

First name: Jim

Last name: Bryant

PIN: ****

Home branch: Headquarters ⌄

Locations

Voice Mail Handling

Privileges
- ☑ Admin
- ☐ Deny configuration
- ☐ Deny call recording
- ☐ Deny forward v-mail to e-mail
- ☐ Hide outside calls

Dialing restriction: <none> ⌄

Call recording
- ☐ Record all calls to mailbox

Recordings to mailbox: My ⌄

Miscellaneous.
- ☐ Save screen layouts when logging out

Outgoing CID Number:

Outgoing CID Name:

OK   Cancel

First name and Last name: Enter the user's first and last names which will be used in the company directory.

PIN: Enter the PIN that the user will supply when logging in. You may want to leave the PIN blank for now. The user will be able to change their own PIN at any time . User's should be encouraged to supply a PIN in order to keep their voice mail secure.

Home branch: Choose the home branch for this user. This will determine the time zone used for determining the date and time for the user's voice mails.

Admin: If you check the box for the Admin privilege, the user will have the Admin and Operator menu items added to their main user interface menu bar.

Deny configuration: If you do not want the user to be able to make changes to their own configuration, check this box.

Deny call recording: If you do not want the user to be able to initiate recording of a call, check this box.

Deny forward v-mail to e-mail: If you do not want the user to be able to forward their voice mail to email, then check this box.

Hide outside calls: If this user's outside calls are to be hidden from other user's call displays, then check this box.

Dialing restriction: If this user's outgoing calls should be restricted, then choose the appropriate dialing restriction from the list. If the dialing restriction is <none> then the user can dial any outside number. Dialing restrictions are defined in sys.local.cfg. When a user is associated with a phone, the user's dialing restriction setting overrides the phone's dialing restriction.

Record all calls to mailbox: If you want all of this user's calls recorded, check this box. If this box is checked.

Recordings to mailbox: If "Record all calls to mailbox" is checked, select the voice mailbox where the user's call recordings should be stored. This list includes "My" and a list of groups. If you select "My", the user's own voice mail folder will be selected. The recordings will be put in the user's or group's new voice mail folder when the user leaves the call or the call ends.

Save screen layouts when logging out: If you check this box, then any changes the user makes to their user interface during a session will be automatically saved when they exit the user interface. This feature should not be used if the user is likely to make undesirable changes to their configuration or window layout and then save them away when exiting the UI.

Outgoing CID number and Outgoing CID name: If you provide values for these settings, they will be used as the outgoing caller ID for the user if the call is going out on a T1 or SIP trunking service. This feature requires the support of the service provider.

Next click on the Locations button. This will present a selection box where you can choose to edit an existing location profile, delete an existing location profile or create a new one:

The "office" location should already be selected. Click on the Edit button to edit this location profile for the new user:

Location office

Location name: office

Voice type: Station

Station number: 110

☐ Flash treated like hangup and pickup

☐ Dialtone after call ends

☑ Auto offhook on intercom calls (requires Polycom or Cisco IP phone)

☑ Page to phone (requires Polycom or Cisco IP phone)

☑ Sounds during instant messaging

Play pages on PC speakers: Never

Speaker volume: 2

Incoming call when you are not on another call

☑ Ring phone   ☑ New call notify box

Incoming call when you are on another call

☐ Call waiting tone   ☑ Call waiting notify box

Notify box ring sound: None

Voice mail

☐ Notify box for new voice mail

New voice mail notification sound: N/A

Minutes between VM sounds: 2

Default status: at desk

OK   Cancel

Location name: This shows the location name you are editing. It cannot be changed. If you want to edit a different location profile, click on the Cancel button, then click on Locations from the User window and select a different location to edit from the Location Selection window or click Add to add a new location.

Voice type: You can select Station or None. Select Station if the user has a FacetPhone station at this location. Select None if when at this location the user will have the FacetPhone graphical user interface but no telephone. None would be also be selected for users who only need a FacetPhone voice mail box.

Station number: This is the number of the physical station (telephone) that the user has at this location.

Flash treated like hangup and pickup: This setting is for users who make many outbound calls and never need to put a call on hold. Hanging up the phone and then quickly picking it back up to place another call can look like a hook flash to the system. If this is not what is intended, then the current call is accidentally put on hold instead of being ended. Turning on this setting causes the system to treat a flash as though the user hungup on the active call and then picked the phone back up to start a new call. This prevents the user from ever being able to put a call on hold using the switch hook or Flash button on the telephone.

Dialtone after call ends: When a call is ended by the other party and the phone is still offhook, it normally leaves the phone "dead". This setting causes the user's phone to be given a new dialtone in that situation so that it is ready to place a new call.

Auto offhook on intercom calls (requires Polycom or Cisco IP phone): This setting is only meaningful if there is a Polycom or Cisco IP phone at this location. If this box is checked, then if someone at another inside extension calls this user, the phone will not ring but will beep and then be taken offhook in speaker phone mode and the caller is immediately connected. If the phone is a Polycom 30x model, the user will not be able to talk back to the caller since it is not a full speaker phone. With any model, you can pick up the receiver and continue the call using the handset.

Page to phone (requires Polycom or Cisco IP phone): FacetPhone has a built-in paging system that allows users to send pages over the IP phones or the PC speakers of other users that are logged in to FacetPhone. This setting is only meaningful if there is a Polycom or Cisco IP phone at this location. If this box is checked, then pages will be played on the phone's speaker when it is on-hook.

Sounds during instant messaging: By checking this box, the PC speakers will make a sound whenever you start an IM session and whenever you send or receive an IM message. This will work only if your PC has sound capability. To turn off IM sounds, uncheck this box.

Play pages on PC speakers: FacetPhone has a built-in paging system that allows users to send pages over the IP phones or the PC speakers of other users that are logged in to FacetPhone. You can choose whether you want pages to be played on your PC speakers at this location. Select "Never" if you don't want any pages played on your PC speakers. Select "When Not on the Phone" if you want pages played over your PC speakers only when you are not using the phone. Select "Always" if you want pages to always be played on your PC speakers, even if you are on the phone. If you have an IP phone and have chosen to have pages played on the phone, then you probably want this set to "Never".

Speaker volume: This setting is used to adjust the volume of pages and voice mails played on your PC speakers in relation to the volume of other sounds that may be played on your speakers.

Incoming call when you are not on another call:

> Ring phone: When this box is checked, the phone specified for this location will ring whenever you receive a call and your phone is idle.

> New call notify box: When this box is checked, the user interface will display a notification window whenever you receive a call and your phone is idle. The notification window will include any available caller ID information of the person calling you. Note that when a call is transferred to you, the outside caller ID information will be displayed. But if the call is parked and the operator is calling you, the operator information will be in the caller ID field.

Incoming call while you are on another call:

Call waiting tone: When this box is checked, you will hear a call waiting tone in the receiver of your phone whenever you receive another call and you are already on a call. To pickup the incoming call you can click on it in the call display and choose activate. This will put your current call on hold and answer the new call that is ringing. You can also press the flash button on your phone to put the current call on hold and answer the call that is ringing. Note that only you will hear the call waiting tone. The

other party on your call will not hear it. The caller ID of the new caller will be presented on the phone along with the presentation of the call waiting tone.

> Call waiting notify box: When this box is checked, the user interface will display a notification box whenever you receive a call and you are already on a call. The notification window will include any available caller ID information.

Notify box ring sound: When selecting the New call notify box or the Call waiting notify box, this drop down list gives you the option of picking from 20 different sounds for an audible indicator on your PC. Or, you can choose None for no ring sound on your PC.

Voice mail:

> Notify box for new voice mail: When this box is checked, the user interface will display a New Voice Mail notification window whenever new voice mail is received.

> Voice mail ring sound: When selecting the Notify box for new voice mail, this drop down list gives you the option of picking from 20 different sounds for an audible indicator on your PC. Or, you can choose None for no ring sound on your PC.

> Minutes between VM sounds: This lets you select how often in minutes to have the audible voice mail indicator played on your PC. This field is grayed out and not changeable if you have not selected a Voice mail ring sound. If you enter 0 in this field, FacetPhone will play the audible indicator only one time without repeating.

Default status: This field lets you set the initial status that you will have when you login at this location.

Different combinations of the call ringing and call notification options make sense depending on how you use FacetPhone.

For operators who must be notified of each incoming call, it probably makes the most sense to have the phone ring when it is idle and have the call notification window be presented for calls when you are already on the phone. The notify box ring sound should be set to an audible value. This way the call notification ringing sound played over the PC speakers while you are already on the phone may be overheard by the current caller making them aware of your need to answer the other call.

Users who want calls to go to voice mail when they are already in a call, should un-check both Call waiting tone and Call waiting notify box options.

Users who want call waiting and use the FacetPhone user interface will probably want to check the Call waiting notify box option. Users who want call waiting but do not use the FacetPhone user interface would need to check the Call waiting tone option.

After completing the configuration of the user's office location profile, click on the OK button to save the settings for that location.

The other location profile you might want to change is the "not logged in" profile. By default this profile has no telephone associated with it. However, you may want your normal phone to ring even when you are not logged in. For this case, edit the "not logged in" location profile and specify your normal station. You must also set the default status for this location to be "at desk" or some other status which shows that you are available for phone calls.

You should now be left with the main user configuration dialog box still displayed. Click on the Voice Mail Handling button. This will present the Voice Mail Handling dialog box:

Voice Mail Handling

Voice mail delivery and notification methods

- ☑ Voice mail to mailbox
- ☐ Voice mail to e-mail: [_____]
- ☐ Notify to e-mail [_____]
- ☐ Notify by phone [_____]
- ☐ Notify to SMS [_____]

From address for e-mails: [_____]

- ☐ Include missed calls in voice mail

User menu script:	user_menu
On-hold script:	<none>
Dialtone Replacement script:	<none>
Idle rings:	4

Generic handling

⦿ All VM callers sent to generic script

Generic script: vm_generic

Selective handling

◯ VM Callers sent to script determined by user's status

No answer script:	vm_idle_park_page
Busy script:	vm_busy
Unavailable script:	vm_unavail
Out script:	vm_out

OK   Cancel

Voice mail delivery and notification methods:

Voice mail to mailbox: Check this box if you want the user's voice mail to be delivered to their voice mail folder on the FacetPhone system. This allows them to listen to their voice mail from their telephone or by calling in from outside. Most people will want to have this option turned on.

Voice mail to e-mail: Check this box if you want the user's voice mail messages sent to their email as wav sound attachments. In the box to the right of this checkbox, enter the email address where the voice mail should be sent. Multiple email addresses may be specified and must be separated by a semi-colon or comma. The total length of this entry must not exceed 1024 characters.

Notify to e-mail: Check this box if you want a notification email sent when the user receives a new voice mail. The notification email includes a subject line with the callerID information, but does not include any email message body. This option does not send the actual voice mail message as a wav attachment. In the box to the right of this checkbox, enter the email address where you want the notifications to be sent. Multiple email addresses may be specified and must be separated by a semi-colon or comma. The total length of this entry must not exceed 1024 characters.

Notify by phone: Check this box if you want the system to call the user when they have a new voice mail. In the box next to this check box enter the phone number(s) that should be called. Phone numbers must begin with the leading 9 or 8 plus trunk group number. Multiple phone numbers may be separated by a comma. When the user receives the notification call, they will be prompted to press * and then will be prompted to enter their user extension and PIN number. At that point they will be in the user menu and can listen to and manage their new voice mail. Note that the system will continue calling until they respond with the *.

Notify to SMS: Check this box if you want the system to send the user an SMS text message to notify them of a new voice mail. In the box next to this checkbox enter their SMS address, usually phonenumber@provider. The text message will contain the same information that an email notification contains. Multiple SMS addresses may be specified and must be separated by a semi-

colon or comma. The total length of this entry must not exceed 1024 characters.

From address for e-mails: This is where you enter the email address to be used as the "From:" address of any emails sent by FacetPhone.

Include missed calls in voice mail: Check this box if you want this user's missed calls to create a voice mail with the caller's information. The audio of the voice mail will only say "missed call". However, the time and caller ID of the caller will be available. Missed call voice mails follow the same delivery and notification rules as any other voice mail.

User Menu Script: This is the IVR script that is run when the user goes into their voice mail management by pressing * on the telephone. By default, this is set to user_menu.

On Hold Script: This is the IVR script that is run when the user puts a caller on hold. If the user parks the caller instead of putting them on hold, the group setting will be used. The options for this are:

<none> -- When None is selected, FacetPhone will use whatever option is selected for the operator group for the user's phone.

message_on_hold - This will play a message to callers when they are put on hold.

music_on_hold - This will play music to callers when they are put on hold.

Dialtone Replacement script: This specifies an IVR script that will be run when the user takes their phone offhook at this location instead of presenting a dialtone. This setting us usually set to <none>.

Idle Rings: This is the number of times FacetPhone will ring the user's phone before sending callers to their voice mail.

Generic handling:

All VM callers sent to generic script: Select this option if you want to treat all callers that go into the user's voice mail the same way. If you want different announcement messages to be played depending on whether the user is on the phone, or out of the office, then do not select this option and see Selective handling below.

Generic script: This shows the name of the IVR script to be run whenever a caller goes into the user's voice mail when generic handling is being used. The default is vm_generic which will play the generic announcement message.

Selective handling:

VM callers sent to script determined by user's status: Select this option if you want a different announcement to play depending upon the conditions described below.

No answer script: This is the script to be used when a caller goes into the user's voice mail after the maximum number of rings have occurred and they have not answered their phone. The default IVR script for this is vm_idle.

Busy script: This is the script to be used when a caller goes into the user's voice mail because their phone is busy. The default IVR script for this is vm_busy.

Unavailable script: This is he script to be used when a caller goes into the user's voice mail because their status indicates that they are not availalbe for a phone call.

Out script: This is the script to be used when a caller goes into the user's voice mail because their FacetPhone status shows that they are not at work. The default IVR script for this is vm_out.

Click OK on the Voice Mail Handling window and then click OK on the main user configuration window.

The new user should appear in the list of users in the administration window.

Repeat this process for each user to be added.

If you want to change any of the settings for the models that you are using to create new users, you can edit the model itself and change it to have the settings you want.

When you are finished, you can click on the close button on the "User Selection" window's title bar [X] to close that window.

Recording Your Own Auto-attendant Prompts

You will want to record your own auto-attendant prompts even if you use the default auto-attendant scripts. To do this, a person with operator or administrator privilege can choose on the main menu:

> Operator / Record Group Voice Prompts

This will present a group selection window:

Only the groups for which you are an operator will be listed. After you choose the group, the prompt selection window will be presented:

Select the prompt you want to record. Click the Record button. If your phone is on-hook when you click record, it will ring. Once you are off-hook follow the directions on the phone to record, play back and save your prompt.

Your FacetPhone installer will instruct you on the prompts that need to be recorded for your auto-attendants.

The prompts that are included in the drop list are specified in the files:

> /usr/facetphone/config/ivr.cfg
> /usr/facetphone/config/ivr.local.cfg

Lines in these files are of the form:

```
<ivr_script=company_closed_menu>
<ivr_prompt=company_closed_greeting>
```

The <ivr_prompt=> lines define the items that are in this drop list. If you need to add prompts to the list, create the ivr.local.cfg file if it does not already exist and add the desired <ivr_prompt=> lines. Any time you edit the ivr.local.cfg file you must signal FacetPhone to re-read the configuration. This is done with the command:

```
facetphone reload
```

Prompts that you record this way are kept in the directory:

/usr/facetphone/ivr/groupname/vmprompts

where groupname is the name of the group that the prompt is for. If it is the default group, the name will be _default_ rather than the name of the default group defined in the sys.local.cfg file. When an IVR script is searching for a prompt, it looks first in the group directory, then the localsystem directory, and finally the system directory. The system directory is updated every time the FacetPhone software is updated, but your group or localsystem prompts will not be changed.

If you would like to have your customized announcements recorded in the standard "voice of FacetPhone", please contact FacetCorp.

Scheduling Your Auto-attendant Operation

You can define scheduled events to turn the auto-attendant on or off or change which auto-attendant is in effect. To define a scheduled event, open the Administration window by choosing on the main menu:

Admin / Change system configuration

On the Administration window choose from the menu:

Add / Add a Scheduled Event

This will display the scheduled event entry window:

Event name: Enter a name for the event that relates to its function, such as Company open.

Hour in 24 hour format and Minute: Enter the hour and minute that the event should occur.

Override other events occuring at the same time: This setting is only used for events that occur on special days such as a holiday. Checking this box will cause the event to override another event that occurs at the same time. For example, a holiday morning event would need to override the normal company open event that would be defined to occur at the same time. Note that an override event will only override other events scheduled for the same time for the same group.

Select by days of the week or Select by days of the year: Normal company open and company closed events should be set as "Select by days of the week". Holiday events should be set as "Select by days of the year".

If "Select by days of the week" is chosen, then check the box for each day of the week that the event should occur. For example, for the normal company open event for a company open Monday through Friday, you would check the Monday, Tuesday, Wednesday, Thursday and Friday boxes.

If "Select by days of the year" is chosen, then enter the days and month that the event will occur. For example, if the company will be closed on December 24th and 25th, then you would enter 24,25 for the days of the month and 12 for the month of the year for a holiday event.

Group: Each auto-attendant event is for a particular group. Choose the group the event refers to from the list of groups.

Auto-attendant on: If this event causes the auto-attendant to be turned on, then check this box. If the event should cause the auto-attendant to be turned off, then uncheck the box.

ON script name: If the event turns the auto-attendant on, then choose the auto-attendant script to be used from the list. The script names in this list come from the ivr.cfg and ivr.local.cfg files. If the auto-attendant is turned off by this event, then the script name is irrelevant but should be set to <none> for clarity.

Most systems will have a company open event that occurs at opening time on business day mornings and a company closed event that occurs at closing time on business day evenings. Holiday events that override the normal events will be defined as needed.

Setting up "Music on Hold" or a "Sold on Hold" Message

If you are using "message_on_hold" for any group's on hold script, then you can record the message to be played to parties on hold or parked by the same method that you recorded your auto-attendant prompts. The prompt in the list named "on_hold_message" is the one to record for this purpose. The message_on_hold script allows the caller to dial 0 to transfer to an operator and 1 to leave voice mail, so you can include this information in your message. If the caller is in this script because they are on hold, then they will go into the voice mail of the person who put them on hold. If they are parked or put on hold at a station that does not have a user, they will go into the group's voice mail. The default on_hold_message prompt is simply the instructions to dial 0 or 1.

The "music_on_hold" script randomly chooses from 10 songs to play. If the first song finishes before the caller is picked back up, then it proceeds through the other songs in order. After playing a song, the script plays "music_on_hold_instruct" to give the caller their options while on hold. The script allows the caller to press 0 to transfer to an operator and 1 to leave voice mail by the same rules for leaving voice mail as with the on_hold_message script. In addition, the caller can press 2 to change the song and 3 to turn the music off or back on.

The system comes with 10 royalty-free songs. To provide your own music, you can override the default songs by putting your music in the ivr/localsystem/vmprompts directory in files named song1.wav, song2.wav, ... song10.wav. The files must be wave files sampled at 8000 Hz, mu-law, mono, 8 bits. We have also found that most music needs its volume reduced to 30% of the original to be the correct volume for the phone.

At FacetCorp, we use a shareware utility named Audacity to convert music files to the correct format. There are undoubtedly many utilities that will do this, but if you don't already have one, Audacity is known to work and is free. The default songs came from www.sounddogs.com and you may search for different or additional royalty-free songs to buy from their large selection.

The Configuration Report

Each time the configuration is changed and each time FacetPhone is started, a summary report of its configuration is written to /usr/facetphone/logs/config_report.txt . This report provides a quick reference to the configuration details of your FacetPhone server. You may view this report from the FacetPhone server web page.

Using Care With the Administrator

Note that when you use the FacetPhone administrator window, you can easily cause stations and lines to be disabled since edit operations on these devices or their gateways cause the device to be disabled until the edit is complete. Under the admin menu, there is an additional item "View System Configuration" that can be used to view the configuration without any danger of disabling devices. No changes to the configuration can be made in this view-only mode.

You Should be Up and Running!

If the above steps went smoothly, your FacetPhone system should be up and running. If not, please give us a call for help.

Have all users login to FacetPhone to make sure their user interface is working properly. Also have each user dial * on their telephone and follow the instructions to record their name for the company directory and to record their voice mail greeting(s).

Next you should look over the sections below that describe various special topics that might apply to your site.

Multiple Instances of FacetPhone on a Server

FacetPhone supports multiple instances running on the same server that is useful for executive suites or hosted services that want to use a single server for multiple companies.

Each instance must have its own IP address, so you must begin by configuring multiple IP addresses for the server.

Next, create the file /etc/facetphonedirlist which will contain the name, installation directory and IP address for each FacetPhone instance:

```
company1        /usr/facetphone1          192.168.10.1
company2        /usr/facetphone2          192.168.20.1
```

When you install FacetPhone you will provide the install.sh script an argument which specifies the instance you want to install:

```
install.sh company1
```

You will then configure each instance independently. Instances may not share the same gateway. Each instance must have its own gateways. All gateways for an instance must have addresses that can be accessed from the intance's IP address on the server.

Backup Server Operation

If you have one or more FacetPhone backup servers, this section explains how to have them takeover and give back control of your FacetPhone system. There are various scenarios depending on the number and configuration of the branches in your system. Some of these will be discussed below. If you have any questions about a backup plan for your system, FacetCorp support can help you develop a plan for your specific configuration.

It is assumed that backup servers are always running and that someone at each site knows how to login as root and enter commands to the shell.

The primary server that runs the FacetPhone system under normal operation is the "master" server. Any other servers are considered "backup" servers and are setup to constantly update their FacetPhone configuration and voice mail files from the master server.

In general, the "facetphone" command is used to start or stop FacetPhone or adjust the branches a master server is controlling.

Scenario 1. One primary server and a backup server at the same branch.

> In this scenario there is only one branch that has servers. The backup server is meant to replace the primary server in the case of a hardware failure.

> To have the backup server takeover, make sure the primary server is no longer running. You can do this with the command:

> facetphone status

> If FacetPhone is running, enter the command:

> facetphone stop

> Then on the backup server enter the command:

> facetphone master

> If you only have one branch defined, it will start the server immediately. If there are multiple branches, it will prompt for each branch whether you want the backup server to control it. You would enter 'y' for each branch in this scenario.

To have the primary server take back over, first have the backup server give up control by entering the command:

facetphone giveback

Then on the primary server enter the command:

facetphone start

Scenario 2. One primary server with local branch backup servers.

In this scenario there is a primary server at the main branch and one or more other branches have local backup servers. One of the backup servers can takeover for master server if the master has a hardware failure. This procedure would be the same as in the scenario above. A local backup server can also takeover only for its local branch if the branch has lost connectivity to the main branch. To do this first have the master branch give up control of the local branch with the command:

facetphone branch

Answer 'y' for each branch the master server will continue to control and 'n' for any branch that will be controlled with its own local backup server.

On the backup server at the local branch enter the command:

facetphone local

This will cause the local backup server takeover only its local branch.

When connectivity to the local branch is restored, on the local branch server enter the command:

facetphone giveback

Then on the master server, take back control of the local branch with the command:

facetphone master

Answer 'y' when prompted whether to control the branch being taken back over by the master server.

After returning from backup operation, any new voice mails left on the backup server must be manually copied back to the master server. Please contact FacetCorp support for help with this step. If possible, please contact FacetCorp support when having a backup server takeover so that we may assist in this process.

FacetPhone Integration with UNIX Applications Using UTAPI (Unix Telephone API)

If you have a UNIX application that includes phone numbers in its data, then you can make minor program changes to enable your users to dial directly from the application. You can also enable the program to select a customer screen ("screen pop") based on the caller ID of an incoming call.

You must be able to run external UNIX programs from your application using a facility such as the "system" C library function in order to modify it to use UTAPI. If you have this capability in your development language, then implementing UTAPI is simple.

Installing UTAPI on Your Application Server

The latest FacetPhone UTAPI software can be obtained from FacetCorp's ftp site:

ftp://ftp.facetcorp.com/facetphone

Choose the correct OS directory for your application server. In that directory, there will be a utapi.cpio file. This is the installation file you need to download.

If you are running the FacetPhone server software on your application server (not recommended except for very, very small systems), then there is no need to install the utapi.cpio file. The UTAPI support is included in the FacetPhone server software installation.

Login as root on the application server. Put the downloaded installation file in a temporary directory. Extract the installation files from the archive with the following command depending on your operating system:

Linux:

    cpio -idumv < utapi.cpio

SCO, AIX (and most others):

    cpio -icdumv < utapi.cpio

Run the utapi_install.sh installation script. Enter the directory where you wish to install UTAPI when prompted. The default is /usr/facetphone_utapi. After the installation is finished the following directories and files will have been created:

/etc/utapidir

   File that contains the path to the UTAPI root directory.

/usr/facetphone_utapi/

   Default UTAPI root directory.

/usr/facetphone_utapi/log.txt

   Log file for the UTAPI install.

/usr/facetphone_utapi/partno.txt

   Current part number and version for UTAPI programs.

/usr/facetphone_utapi/bin

   Directory containing the UTAPI programs.

/usr/facetphone_utapi/bin/fp_dial

   Program your application will run to dial a telephone number.

/usr/facetphone_utapi/bin/fp_cid

   Program your application calls to check for caller ID information.

/usr/facetphone_utapi/bin/fp_app

   Sample application program to use with the fp_srvr program to test the correct setup of the fp_utapi.cfg file.

/usr/facetphone_utapi/bin/fp_srvr

   Sample server program to use with fp_app for testing the correct setup of the fp_utapi.cfg file.

/usr/facetphone_utapi/config

   UTAPI configuration directory.

/usr/facetphone_utapi/config/fp_utapi.cfg

> Configuration file used by the UTAPI programs.

/usr/facetphone_utapi/source

> Source directory for fp_app.

/usr/facetphone_utapi/source/app_ux.c

> Source for fp_app program. Included to show what system calls your application will need to make to call the fp_cid and fp_dial programs.

Configuring UTAPI

Edit the /usr/facetphone_utapi/config/fp_utapi.cfg file and set the <fp_addr=> parameter to the IP address or DNS name of the computer on which the FacetPhone server software is installed. Set the <fp_port=> parameter to the number used by the FacetPhone server software (default is 6500).

If your FacetPhone system is running and you are logged in to the FacetPhone user interface, then you can test the UTAPI configuration by running the UTAPI dialing program manually:

> /usr/facetphone_utapi/bin/fp_dial -u user -n 9,9728441234

where you substitute your user name after the -u switch and provide whatever number you want to dial after the -n switch. Be sure to include the leading 9 (and optional comma) to indicate that you are making an outside call.

Modifying Your Application to Do UTAPI Dialing

Using whatever method your application development language uses to run an external UNIX application, run the fp_dial program:

> fp_dial -u *username* -n *number*

where *username* is the name of the user running your application who is logged in to the FacetPhone system. It is therefore important to have the FacetPhone user names match the regular computer login names to avoid having to map one user name to another. The *number* must include the leading 9 for an outgoing line or alternate trunk access codes if desired.

Users must be logged in to the FacetPhone user interface in order for the system to know which phone they are using. If the user's phone is on-hook, it will ring. If it is off-hook, the call will start. The fp_dial program does not return any response or acknowledgment back to your application. The file:

/usr/facetphone_utapi/source/app_ux.c

shows how this call is made if your application is written in C.

The only other task for you is to provide a way within your application for your users to identify a number they want to call and issue a command to do so. For example, the FacetCorp customer database has been modified to allow us to press a function key and pop up a dialing box where we can step through the phone numbers associated with a customer record and press D to dial the current one.

Modifying Your Application to Use UTAPI Caller ID Information

The UTAPI caller ID program reports the caller ID information for the call that is currently ringing the user's phone or the last one that rang it if it isn't currently ringing. This is done by running the fp_cid program from your application:

fp_cid -u *username*

where *username* is the name of the user running your application and who is logged in to the FacetPhone system using the same login name. If caller ID information is available, fp_cid will output the information to stdout. Your application must be able to read the standard output from a program that it runs. If it cannot, you can write a shell script to write the standard output to a file that your application can read when the program finishes. The format of the information is:

<number=9729859901><name=FACETCORP>

If no caller ID is available, fp_cid will write the string:

<number=No caller id>

Users must be logged in to the FacetPhone user interface in order for the system to know which phone they are using. The file:

/usr/facetphone_utapi/source/app_ux.c

shows how this call is made if your application is written in C.

Integration with Windows Applications using TAPI and the FacetPhone Caller ID Program

FacetPhone uses Microsoft's TAPI (Telephony API) interface. All Microsoft Windows platforms come with TAPI installed. This API allows applications to interact with telephony and other communication systems to make and receive telephone calls.

FacetPhone TAPI Architecture

FacetPhone includes a TAPI Service Provider (TSP) which provides the interface between the Windows TAPI Server program (TAPISRV) and the FacetPhone system. In addition, FacetPhone provides a caller ID program to deliver caller ID data from incoming calls to some popular applications that do not use TAPI for this purpose. The FacetPhone User Interface (UI) mediates the communication between the FacetPhone server and the TAPI service provider and caller ID program. The architecture for FacetPhone integration with Windows applications is shown below.
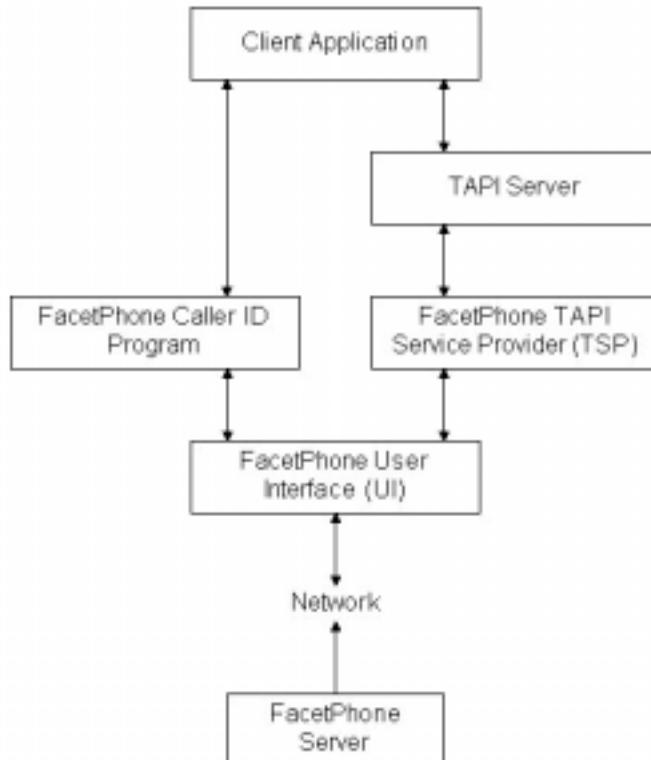
For outgoing telephone calls, the client application sends a dial request to the TAPI server. The TAPI server passes the request on to the FacetPhone TSP DLL. The TSP passes the information to the FacetPhone UI, which sends the request to the FacetPhone server to make the call. Status information is passed back to the client application.

For incoming telephone calls, the FacetPhone server will pass the caller ID information to the FacetPhone UI which will send it to both the FacetPhone Caller ID program, and the FacetPhone TSP. The TSP passes the information to the TAPI server, which then passes the information to the client application.

For those applications that do not use TAPI for incoming telephone calls, FacetPhone includes a caller ID program, fp_callerid, which supplies caller ID information in an application specific manner to those applications. Currently the only applications that are supported by the caller ID program are Microsoft's Outlook (Win2000 or later) and Goldmine.

Configuring the FacetPhone TSP

To configure the FacetPhone TSP, open the Windows Control Panel and double click on the icon labeled "Phone and Modem Options" or "Telephony":
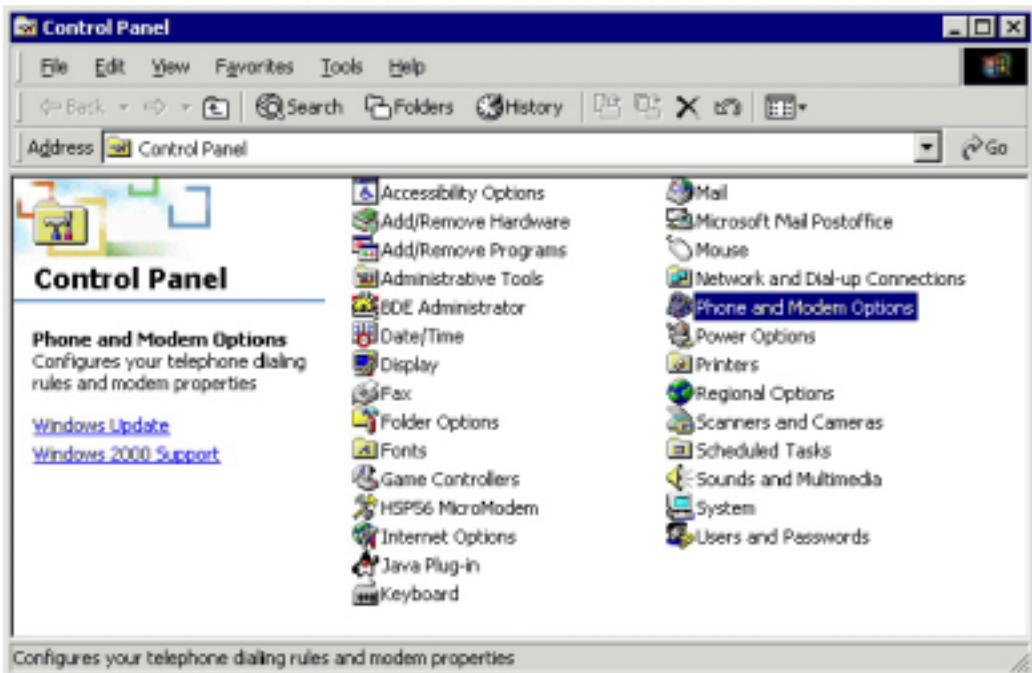
The following example shows the procedure for installing on Windows 2000 Profes-
sional. For earlier versions, after clicking on the Telephony icon in the Control Panel,
you will see a dialog box labeled "Dialing Properties". Click on the "Telephone Drivers"
tab then follow the instructions shown in the "Add Provider" dialog box below. On
Windows 2000 or later, the "Phone and Modem Options" dialog box will be displayed:
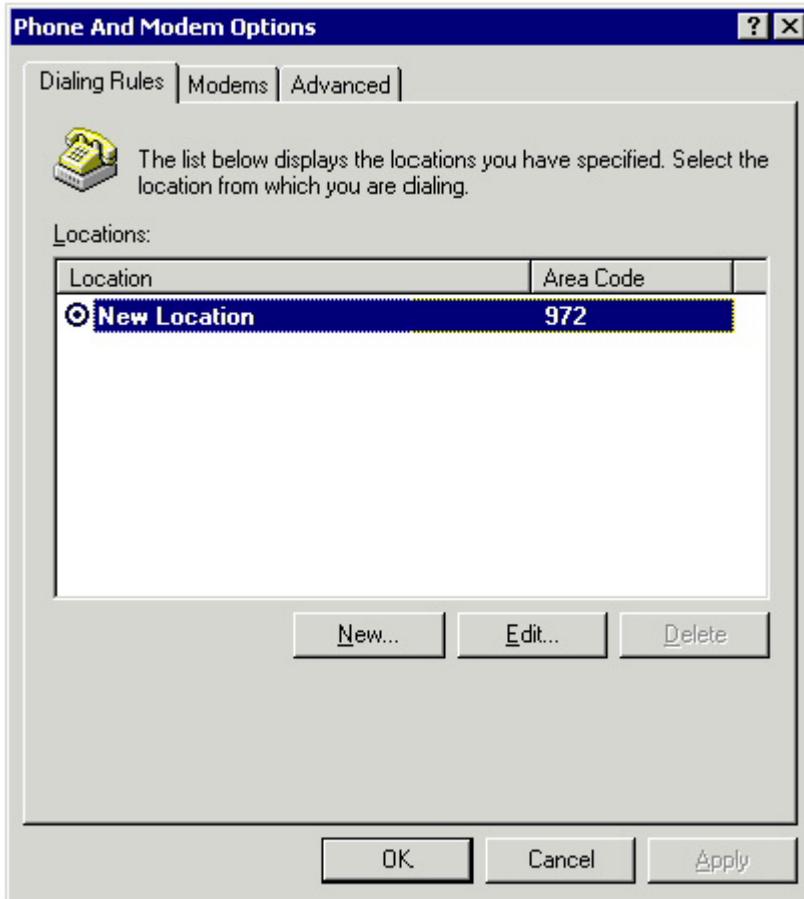
Click on the Advanced tab:
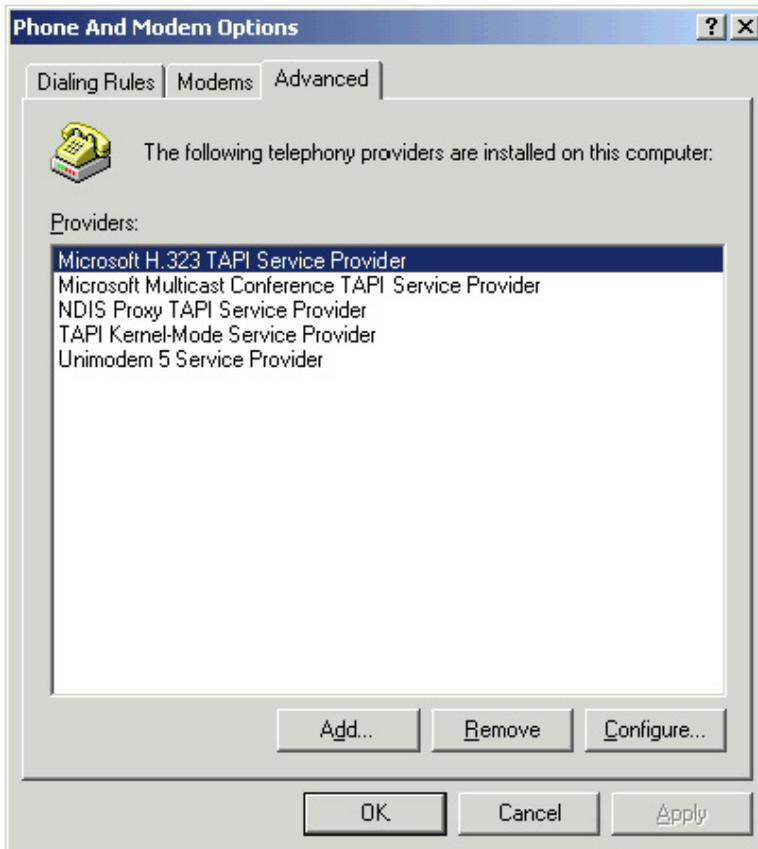
This dialog box shows the TSPs that are currently installed. Click on the Add button and the "Add Provider" dialog box will be displayed:
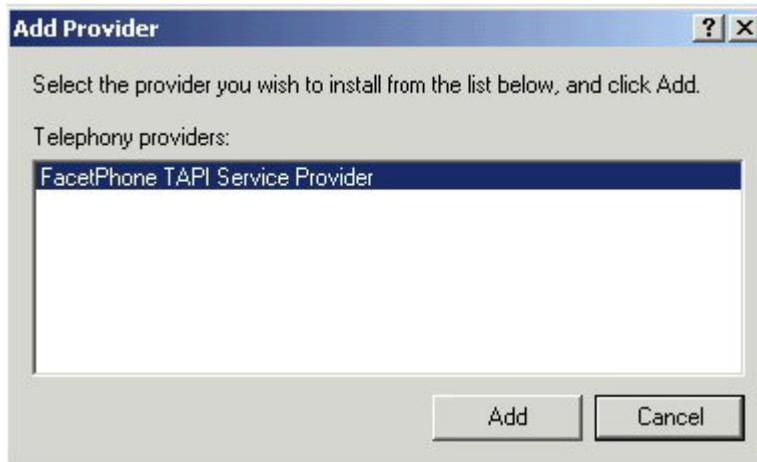
If you do not have an item named "FacetPhone TAPI Service Provider" in the list, then the FacetPhone TAPI software is not installed on this PC. See the section "Installation and Initial Configuration", "Windows PC Installation" above.

Select the FacetPhone TAPI Service Provider item in the list and click the Add button. The "FacetPhone TAPI Service Provider/Caller ID Configuration" dialog box will be displayed:

Unless told otherwise by FacetCorp support staff, keep the default settings for UDP Dialing Address, UDP Dialing Port, UDP TSP Response Port, and UDP CID Response Port.

If you are using an application that can accept incoming caller ID data via TAPI, such as Act!, then check the "TAPI" box in the "Caller ID Notification" section.

If you wish to receive caller ID information on incoming phone calls and you are using either MS Outlook or Goldmine, check the "OTHER" box and enter the appropriate program name in the "App" edit field. If you want new callers to be added to your database, check the "Add New Callers" box.

Currently the only valid values for the "App" field are "Outlook" and "Goldmine". The caller ID program can only communicate with one of these programs at a time. It cannot be configured to communicate with both of them simultaneously.

After all data has been entered on this dialog box, click on the OK button. This will bring you back to the "Phone And Modem Options" dialog box. The "FacetPhone TAPI Service Provider" entry should now be in the list of Providers:

Phone And Modem Options   ? ✕

Dialing Rules | Modems | Advanced

The following telephony providers are installed on this computer:

Providers:

FacetPhone TAPI Service Provider
Microsoft H.323 TAPI Service Provider
Microsoft Multicast Conference TAPI Service Provider
NDIS Proxy TAPI Service Provider
TAPI Kernel-Mode Service Provider
Unimodem 5 Service Provider

Add...   Remove   Configure...

Close   Cancel   Apply

Click on the Close button. Exit the Control Panel.

Starting and Stopping the FacetPhone Caller ID Program

After installing the FacetPhone caller ID program, and rebooting the PC, it should automatically run and will put its icon in the system tray (the telephone icon on the far right):


If you need to manually start the caller ID program click on Start->Programs->FacetPhone->Caller Id . This should cause the icon to be displayed in the Windows system tray.

To terminate the caller ID program, right click on its icon in the system tray and choose "Exit" from the menu that is presented.

Configuring Microsoft Outlook to Dial Using FacetPhone

Outlook can be configured to use FacetPhone when dialing out of its Contacts list. The following screens are from Outlook 2000. Run Outlook and click on the Contacts icon:
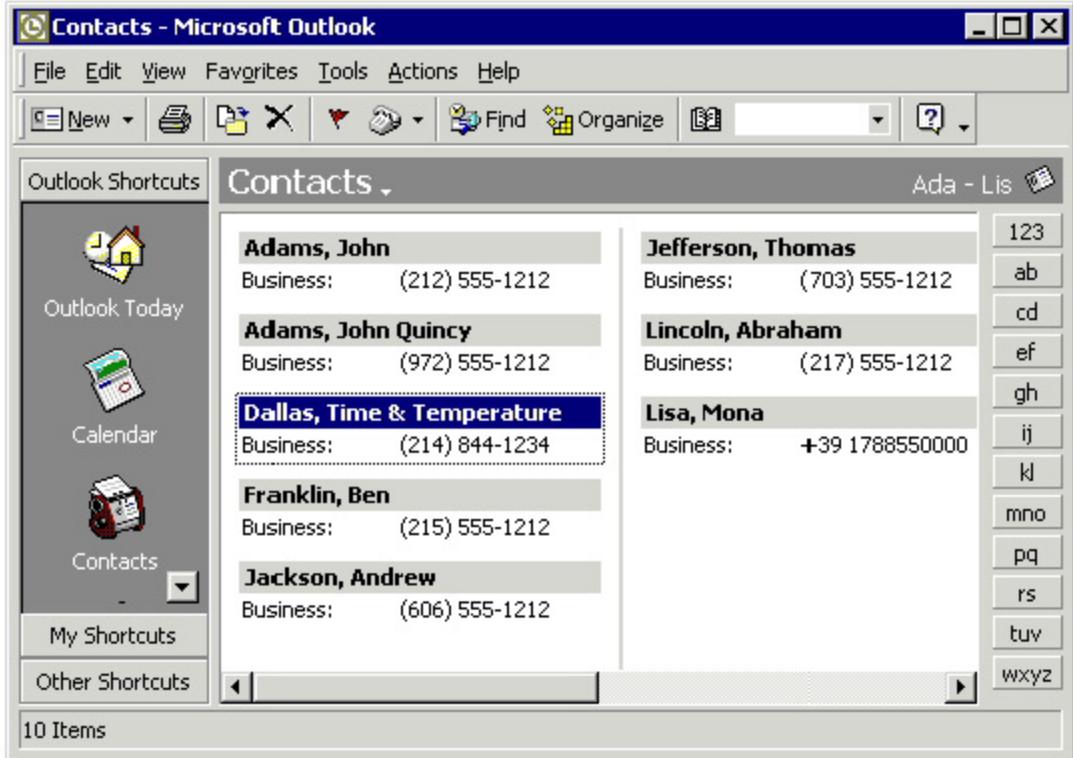
On the menu choose Actions / Call Contact / New Call . The New Call dialog box will be displayed:
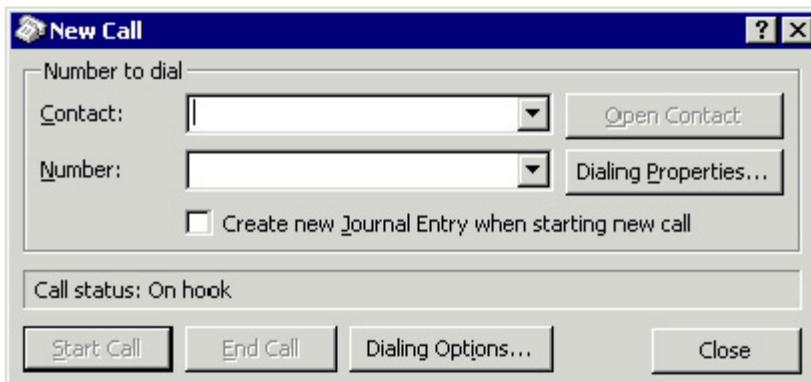
Click on the Dialing Options button and the Dialing Options dialog box will be displayed:

If "FacetPhone Line" does not appear in the "Connect using line" box, click on the pull down list and select FacetPhone Line. You might need to scroll down the list to find it. The Line Properties button is used to access the FacetPhone TSP options (see Configuring the FacetPhone TSP). Click on the OK button then click on the Close button in the New Call screen. Outlook is now setup to use FacetPhone when dialing from your contacts list.

## *Configuring the Windows Phone Dialer Application to Use FacetPhone*

If your version of Windows has the Phone Dialer application, it can be configured to let you use the Phone Dialer with FacetPhone. To run Phone Dialer, on the Start menu choose:

Start->Programs->Accessories->Communications->Phone Dialer.

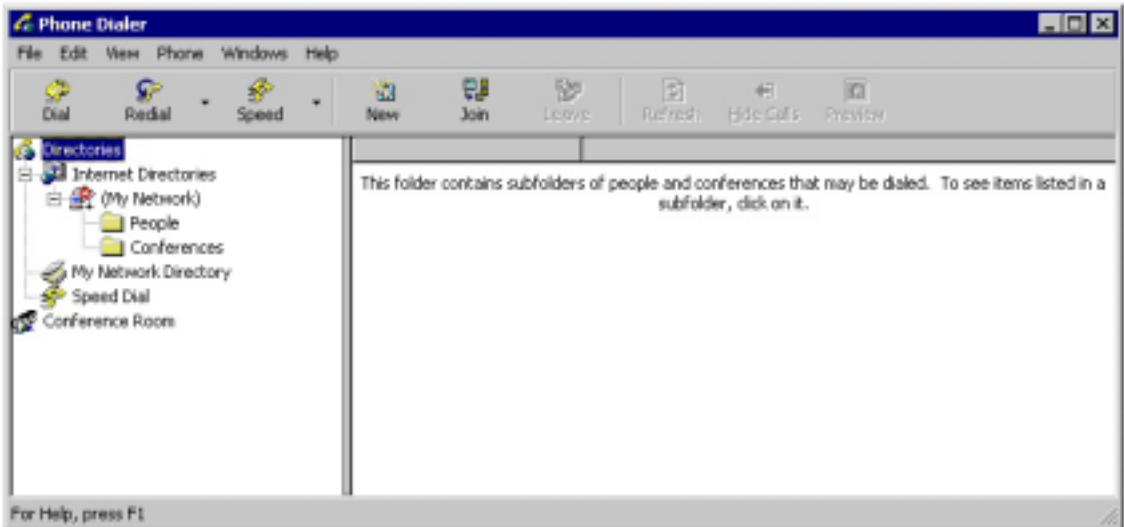The Phone Dialer will be displayed. On older versions of Windows, you will see:

On the menu choose Tools / Connect Using... . The Connect Using dialog box will be displayed. If the Line is not set to "FacetPhone Line", then click on the drop list and choose FacetPhone Line:
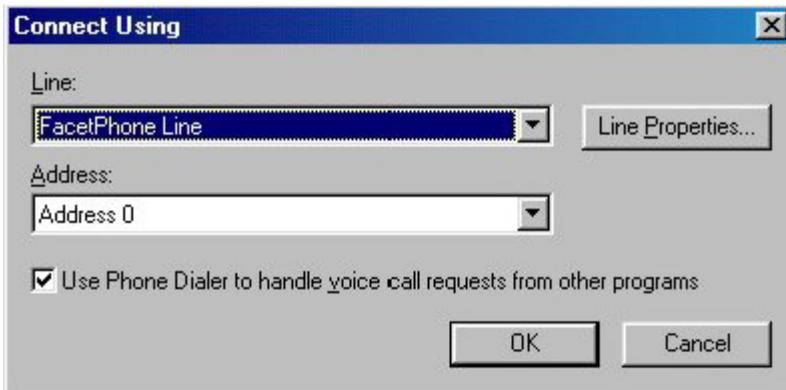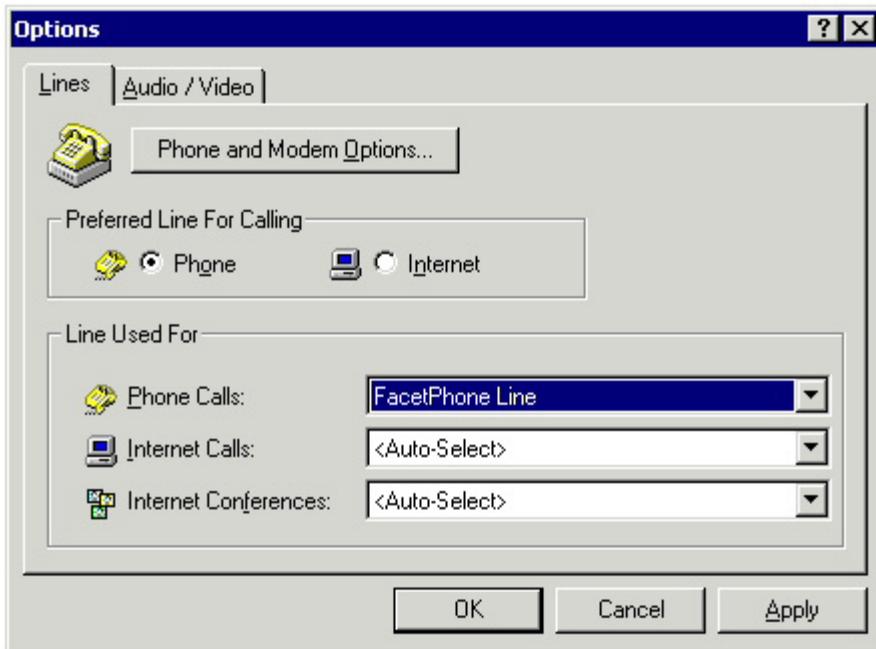
Click OK to save your changes.

On newer versions of Windows, when the Phone Dialer is run, you will see:

Select Directories in the left pane. On the menu, choose Edit / Options... . The Options dialog box will be displayed. Choose "Phone" as the Preferred Line for Calling. In the "Line Used For" section of the dialog box, choose "FacetPhone Line" as the value for Phone Calls:

Click OK to close the Options dialog box.

Dialing from the FacetWin Terminal Emulator

If your site uses the FacetWin terminal emulator, then the UNIX applications that are run in the emulator are already FacetPhone dialing enabled. This requires FacetWin Build 457 or later. The only configuration required to enable this integration is to enable these features in any terminal emulation session configuration you want. Modify the properties of the session and click on the Advanced tab:
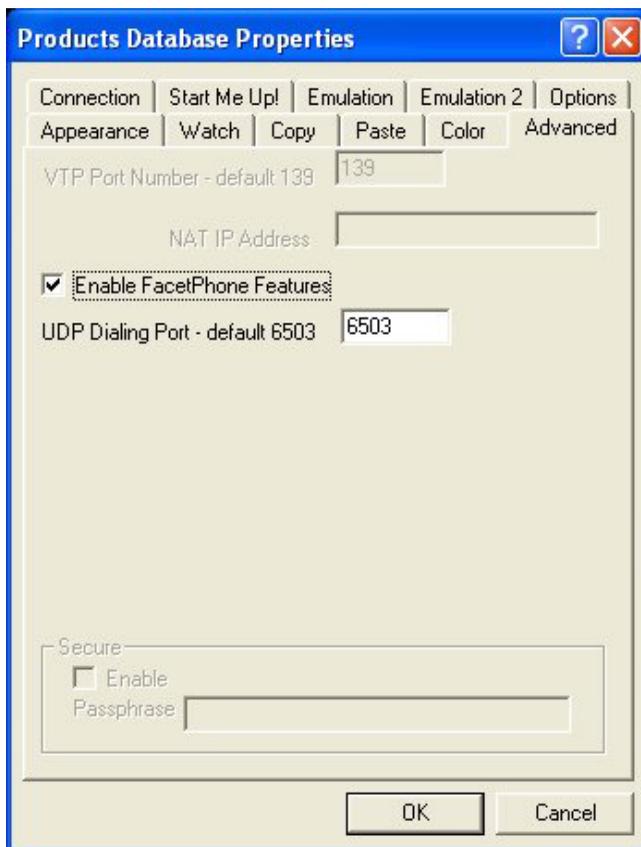
Check the box "Enable FacetPhone Features". Now the user can dial from that FacetWin session when they are logged in to the FacetPhone user interface on the same PC. By simply highlighting a phone number on the screen with the mouse, the user can right click the highlighted area and choose the dial command from the menu. In addition, predefined fields that contain phone numbers can be defined so that the user does not need to highlight the phone number before right clicking on it. Complete use of the dialing features are documented in the FacetPhone User's Guide.