FacetPhone Administrator's Guide

Facet**Corp**

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Introduction

Before reading this guide you should have already read the "FacetPhone User's Guide" for a basic understanding of FacetPhone and the "FacetPhone Manager's Guide" for an understanding of the features you may be asked to configure.

FacetPhone Configuration Files

Most of the FacetPhone configuration is done with the FacetPhone Administrator function in the FacetPhone user interface. However, some of the basic configuration is stored in text configuration files that are manually edited in order to make changes. FacetCorp support staff usually make all the changes to these files. They are documented here so that you know what options are available but we ask that you contact us for help when needing to change them. The default location for FacetPhone is:

/usr/facetphone

This can be modified by editing the /etc/facetphonedir file which contains a single setting:

FACETPHONEDIR=/usr/facetphone

Change the value of FACETPHONEDIR to relocate the FacetPhone installation to another directory. The default location of all FacetPhone configuration files is:

/usr/facetphone/config

All of the FacetPhone configuration files (even those modified through the Administration user interface) are text files with lines of one or more settings of the form:

<dataname=datavalue>

For example:

<default_group=FacetCorp>

You can have multiple settings on one line such as:

<user_low=100><user_high=199>

There must be no spaces between the two settings.

The configuration files that are manually edited are:

sys.local.cfg and sys.branch.cfg – basic system-wide settings.

branch.cfg – definition of each branch.

trunk_groups.cfg – definition of each trunk group (for lines).

sys_speed_dial.cfg – list of system speed dial numbers.

call_routing.cfg – definition of outgoing call routing rules.

ivr.local.cfg – list of available IVR script and prompt names.

skills.cfg – list of available skills to assign to users.

The configuration files that are modified via the user interface and should never be manually edited are:

gateways.cfg – definition of each media gateway and part of the definition of each IP phone.

phones.cfg – definition of each phone on the system.

lines.cfg – definition of each line on the system.

userid.cfg – assignment of user extensions to user names.

users/*.cfg – each user's settings are in a *username*.cfg file in the users subdirectory.

groups.cfg – definition of each group.

schedule.cfg – definition of scheduled events to change auto-attendants.

The FacetPhone User Interface Administration Function

The FacetPhone user interface provides an administration function for those users who have the administration privilege. On the main menu for the user interface there is an Admin item. The sub menu items under Admin are:

Change system configuration

This option will open the FacetPhone Administration window:

This window has its own menu. The Configure and Add menu items will be discussed with the discussion of each of the administration windows below.

View system configuration

This option will also open the FacetPhone Administration window except that the only options you will have is to view information and not change it. Only one person can be in the Change system configuration function at a time but multiple people can view the configuration at the same time.

Open alert messages window

This option will cause a window to be added to your main user interface window that will display any alert messages sent by the servers.

Restart FacetPhone servers and Shutdown FacetPhone servers

These options do what they say. You will be prompted to click OK before the restart or shutdown proceeds. The system can also be shutdown or restarted with the facetphone command described later.

The following chapters will describe each of the configuration files that must be manually edited and then each of the items that can be configured from the user interface administration function.

System-Wide Settings in sys.local.cfg and sys.branch.cfg

The most basic system-wide 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 your FacetPhone software is upgraded. You should never make changes to this file as they will be lost the next time the FacetPhone software is updated.

The sys.local.cfg file is where you specify any settings that are unique for your system 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. No other settings should be in the sys.branch.cfg file. The settings in the sys.branch.cfg file override settings in sys.cfg and sys.local.cfg. When a backup server is syncing with the master server, it does not sync the sys.branch.cfg file since it has its own unique settings that are different from the master's sys.branch.cfg file.

When making a change to any of these settings while the system is running, you can usually have the system reload the settings and they will take effect. From a shell prompt logged in as root run:

>facetphone reload

This will cause most setting changes to take effect immediately. In the description of each setting below, it will note if a setting requires a restart of FacetPhone to take effect.

The following are sys.local.cfg settings. The few settings which should go in sys.branch.cfg instead are noted.

System Names and Addresses

<sysname=FacetCorp>

This is the system name, usually configured with the name of the company. The system name is not currently used in any way other than an easy way for FacetCorp support staff to identify a system. It is reported in the Configuration Report.

<default_group=Company>

The system comes with one pre-defined group. The actual name of the group is _default_ but the default_group setting associates a name with it. Usually the company name is used here. You will probably want to change this from its default value of "Company". A reload while FacetPhone is running will cause a change in this setting to immediately take effect but user interfaces that are connected must logout and login to see the new name on window titles.

<ca_addr=199.29.166.41>

This setting is the call agent's IP address. This is the address that all the IP phones and media gateway devices will use to communicate with the server. This value must always be set. FacetPhone must be restarted for a change of this setting to take effect. If there are multiple branches with servers, then this setting must go in the sys.branch.cfg file instead of sys.local.cfg.

<this_branch=1>

This setting specifies which branch that is defined in the branches.cfg file is the one where this server resides. This is only required if your system has more than one branch. If

there are multiple branches with servers, then this setting must go in the sys.branch.cfg file instead of sys.local.cfg.

<htdocs_dir=/Library/WebServer/Documents>

This setting is the full path to the web server html documents on the FacetPhone server.

<mail_server=127.0.0.1:25>

This setting is the IP address and port where mail should be sent for delivery. This must be properly configured and the recipient system configured to accept the mail in order for voice mail or notifications to be sent to a user by email.

<tftp_dir=/private/tftpboot>

This setting gives the full path to the directory where files are stored on the system for tftp file transfer. Some of the phones or gateway devices use this method to download their configuration files.

<polycom_dir=/ Users/PlcmSpIp/>

This setting gives the full path to the directory where Polycom phones will access their configuration files via ftp.

<mysql_server=199.29.166.41>

This setting is optional and specifies the address where a MySQL server is running to accept FacetPhone call records. By default the FacetPhone server is the location of the MySQL database.

<ca_sip_port=5060>

There are three SIP ports that you can define in order to use alternate ports with SIP trunking services or SIP phones that are outside the company LAN. If you do not specify ca_sip_port, in sys.local.cfg, then its value will be the standard SIP port of 5060.

<ca_sip_port_2=15060>

> This setting specifies a port for the second possible SIP port that can be used. If this is not specified in sys.local.cfg, then it will have a value of 15060.

<ca_sip_port_3=15070>

> This setting specifies a port for the third possible SIP port that can be used. If this is not specified in sys.local.cfg , then it will have a value of 15070.

User, Phone, Line and Dial Plan Numbering

<user_low=100><user_high=199>

> These settings define the range of user extensions. All extensions in the range must have the same number of digits. The default range is 100-199. FacetPhone must be restarted for changes in these settings to take effect.

<phone_low=100><phone_high=199>

> These settings define the range of station numbers. This range should be the same as the user_low and user_high range. FacetPhone must be restarted for changes in these settings to take effect.

> group for an outside call (usually 8). All speed dial numbers should have the same number of digits. By default, speed dials are turned off with speed_dial_low and speed_dial_high set to -1. FacetPhone must be restarted for changes in these settings to take effect.

<line_low=1><line_high=50>

> These settings define the range of valid line numbers. Usually line_low is set to 1 and line_high is set high enough to accommodate the maximum number of lines the system is expected to have connected to it. The defaults are 1 to 50. FacetPhone must be restarted for changes in these settings to take effect.

<nbr_trunk_grp_digits=1>

This setting specifies how many digits must be dialed to specify a trunk group. The default is 1 but if you have trunk groups that go beyond 0-9, then you will need to have 2 digit trunk groups. FacetPhone must be restarted for changes in this setting to take effect.

<speed_dial_low=20><speed_dial_high=40>

These settings define the range of speed dial numbers. You can define any range as long as a speed dial number does not begin with the same digit as an extension number, the digit to make an outside call (usually 9) or the digit to select a trunk.

<dial_oper_prefix=0>

This setting defines what digit on a phone is dialed to ring the phone's operator group. The default is 0 and is almost never changed.

<out_line_prefix=9>

This setting defines what digit on a phone is dialed to access an outside line on the default trunk group for the phone. To dial an outside number, the user would dial this digit followed by the outside number. The default is 9 and is almost never changed.

<out_trunk_prefix=8>

This setting defines what digit on a phone is dialed to access an outside line in a specified trunk group. To dial an outside number on a line in a specific trunk group, the user would dial this digit, followed by the trunk group number, followed by the outside number. The default is 8 and is almost never changed.

System Maximums

<max_branches=50>

This setting defines the maximum number of entries that can be defined in the branch.cfg file. The default is 50 and usually does not need to be specified in sys.local.cfg. FacetPhone must be restarted for changes in this setting to take effect.

<max_group=30>

This setting defines the maximum number of groups that can be created. The default is 30. FacetPhone must be restarted for changes in this setting to take effect.

<max_pub_ustat=20>

This setting defines the maximum number of public user status definitions that can be created. The default is 20. FacetPhone must be restarted for changes in this setting to take effect.

<max_gw=50>

This setting defines the maximum number of media gateways that can be defined. The default is 50. FacetPhone must be restarted for changes in this setting to take effect.

<max_call=100>

This setting defines the maximum number of concurrent calls the FacetPhone system can uspport. The default is 100. FacetPhone must be restarted for changes in this setting to take effect.

<max_text_chat=100>

This setting defines the maximum number of concurrent instant messaging sessions the system can support. FacetPhone must be restarted for changes in this setting to take effect.

<max_nettapi_clients=100>

> This setting defines the maximum number of clients that may connect using the NetTAPI protocol. This is only required if you use software that integrates with FacetPhone using this protocol.

<max_parties_per_call=50>

> This setting defines the maximum number of parties who can be in a call. The default is 50. FacetPhone must be restarted for changes in this setting to take effect.

<max_profiles_per_user=20>

> This setting defines the maximum number of location profiles each user can have. The default is 20. FacetPhone must be restarted for changes in this setting to take effect.

<nbr_job_code_digits=4>

> This setting defines how many digits there are in a job code. Job codes can be associated with calls using the user interface or by requiring the user to dial a job code before placing an outgoing call.

Miscellaneous

<inside_ring_type=1>

> This setting defines the distinctive ring that will be used when a call from another extension is calling. Distinctive rings for outside calls ringing in are definable per group and will be discussed with group configuration below. The default value is 1 which is a double ring.

<allow_monitoring=n>

> This setting is the global setting determining whether monitoring is allowed on the system. The default value is 'n' so that monitoring is disabled. To enable monitoring, this value must be change to 'y' and FacetPhone must be restarted. After

that, then monitoring can be enabled per group as discussed with group configuration below.

<use_full_name_in_user_displays=n>

This setting determines whether the first name and last name of users will be displayed in user windows instead of each user's login name. The default value is 'n' meaning that the login names will be displayed.

<full_name_last_name_first=n>

This setting only has an effect when the setting above is set to 'y' and first and last names are being shown in user windows. If the value of this setting is 'y', then the full names will be displayed with users' last names first. The default value is 'n' meaning that full names will be displayed as first name followed by last name.

< park_not_hold_active_when_picking_up_new_call=n>

This setting determines what happens when you are in a call and pickup another call with the user interface. Normally your first call is put on hold but if the value of this setting is 'y', then your current active call will be parked when you pickup another call with the user interface.

<max_vm_minutes=60>

This setting determines the maximum length of a voice mail. After this maximum is reached, the system will hang up on the caller and end the voice mail recording. The default value is 60 minutes.

<excessive_call_minutes=0>

This setting determines the duration of a call at which the duration in the calls window will be displayed in red. The default is 0 meaning that the call duration will never be displayed in red.

<excessive_hold_minutes=0>

This setting determines the total amount of time on hold, parked or queued before the hold duration in the calls window will be displayed in red. The default is 0 meaning that the hold time of calls will never be displayed in red.

<notify_new_ui=n>

This setting determines whether users will receive a message when they login with the FacetPhone user interface if there is a newer user interface available than the one they are logging in with. The default is 'n' meaning that they will not receive messages about the version of their user interface.

<ustat=name><at_work=y><avail_phone=y><avail_text=y><avail_acd=n>

This setting defines a system-wide user status. The value after "ustat" defines the name of the status as it will appear in users' pull down list on their user interfaces. The avail_work value is 'y' if the user is considered to be at work when in this status or 'n' if they are not. The avail_phone value is 'y' if they are available for phone calls while in this status. The avail_text value is 'y' if they are available for instant messaging sessions when in this status. The avail_acd value is 'y' if they are available to receive operator calls when in this status.

<remove_ustat=name>

This setting is used in sys.local.cfg to remove a standard system-wide setting that is defined in the sys.cfg configuration file. The value after remove_ustat should be the name of the user status that you want to remove.

<dial_restriction=no LD><allow=1800><allow=1888><allow=1877><allow=1866><deny=1>

This setting is used to define a dialing restriction. The value after dial_restriction is the name of the restriction as it will appear in drop lists where you assign a dialing restriction to a user or a phone. The restriction is evaluated from left to right until it finds an allow or deny value that matches the number being dialed. In the example above, numbers that start with 1800, 1888, 1877, or 1866 will be allowed. Any other numbers

beginning with 1 (and therefore dialing a long distance call) will be denied. The caller will be told they cannot dial that number.

<send_queued_call_notification=y>

This setting enables a queued call notification feature. The default value is 'n'. If it is set to 'y', then any operator that has user interface call notifications turned on will also get a call notification window when they are on a call but a new call has been queued to their operator group. The queued call notification window has the caller ID of the queued caller and buttons to pickup the queued call or close the window.

<call_display_when_ui_park=n>

This setting determines whether a call display window will be presented to a user when they park a call with the user interface. If the value of this setting is 'y' then this will be done. This allows the user to enter a note for the call which will be displayed in the calls window after the call is parked. Users can use this to give other users some information about the parked call on their calls display. The default value is 'n'.

<disallow_change_other_user_stat=n>

By default, any user with operator or administrator privileges can change any other user's status using the user interface. This setting can be used to disable the feature. The default value is 'n' meaning that operators and administrators can change another user's status.

<allow_record_all_outgoing=y>

This setting determines whether a user's "Record all calls" setting can cause outgoing calls to be recorded. The default value is 'y'.

<allow_record_all_incoming=y>

This setting determines whether a user's "Record all calls" setting can cause incoming calls to be recorded. The default value is 'y'. If users are answering incoming calls to a group

that is recording all calls, setting this to 'n' will prevent having the user make a duplicate recording of the same call.

<allow_record_all_inside=y>

This setting determines whether a user's "Record all calls" setting can cause extension to extension calls to be recorded. The default value is 'y'.

<encrypt_voice_mail=n>

This setting determines whether voice mail will be encrypted on the FacetPhone system. The default is 'n' and voice mail is not encrypted.

<encrypt_call_recording=n>

This setting determines whether call recordings will be encrypted on the FacetPhone system. The default is 'n' and call recordings are not encrypted.

<write_cdr_to_mysql=n>

This setting determines whether or not call detail records will be written to a MySQL database. The default is to not write to the database. Change the value to 'y' to start adding call records to the database.

<allow_IM=y>

This setting determines whether or not instant messaging is enabled on the FacetPhone system. The default value is 'y' so that instant messaging is enabled.

<write_to_IM_log=y>

This setting determines whether or not instant messaging sessions are logged to the file /usr/facetphone/logs/IM_log.txt. The default value is 'y' and IM sessions are logged.

<acd_oper_change_stat_no_answer><new=away from desk>

This setting is used when automatic call distribution is in use and a call rings to an operator whose status indicates they are

available for operator calls but they don't answer the call. In order to prevent further calls being sent to this operator until they confirm they are ready again, this setting will cause their status to be changed to the user status specified as the new status.

<outbound_agent_change_stat_no_call><old=outbound><new=away_from_desk><minutes=2>

This setting is used to measure the time that an agent spends between outbound calls when in the status specified in the old= setting. If the time between calls exceeds the number of minutes specified in the minutes= setting, then the user's status will be changed to the value specified in the new= setting.

<local_7_digit_dialing=y>

This setting is used to specify that 7 digit local dialing is in effect for the site. This allows the configuration files created for SIP phones to have the proper digit map.

<disallow_oper_priv_to_opers=y>

This setting is used to prevent sending the operator privilege to an operator's user interface. This means they will not have the Operator menu on their UI and they will not have the Reset option in user, line, phone and call menus. They will still receive operator calls. The default value is 'n'.

Defining Branches in branch.cfg

A branch is a physical location where all or part of your phone system is located. You will always define at least one branch in the branch.cfg file for the company headquarters. If you have offices using the phone system in other locations that are connected via a VPN or MPLS, then you would define each of them in the branch.cfg file also. You may optionally want to define single telecommuter locations as branches also. After changing this file, from a shell prompt logged in as root run the command:

> facetphone reload

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

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

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

<branch=0>

> This is the branch number. The headquarters branch should be 0. You can have as many branches as are configured by the max_branches setting in sys.local.cfg. Each line in this file must begin with the branch item.

<name=Dallas>

> This is a descriptive name that should indicate the location of the branch.

<server_type=active>

> 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=199.29.166.34>

This specifies the IP address of the server at the branch, if any. If there is no server, the proper specification is <server_addr=>.

<ctrl_branch=0>

This setting specifies which branch has the server that is controlling the branch being defined. Under normal operations 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=CST6CDT>

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 trunk_groups.cfg

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. Usually you group lines from a similar source together into a single trunk group. For example, the 23 lines that represent the channels on a T1 PRI would be put in the same trunk group. After changing this file, from a shell prompt logged in as root run the command:

      facetphone reload

At a minimum, each trunk group entry includes the trunk group number and a description of the trunk group:

<trunk_group=0><desc=analog lines>

If the lines in the trunk group are on a T1 PRI or are SIP trunks, then they will support DIDs (Direct Inward Dial numbers) allowing callers to dial directly to a group, user or station. You can specify how each DID is routed in the trunk group entry:

<trunk_group=0><did=2146236944><type=user><dest=144>

The value of the did item is the DID as it is delivered from the phone company. It should match the number of digits the phone company is delivering as DID. In the above example if the phone company was only delivering 4 digits, the item would be <did=6944>.

The type can be "user", "group" or "station". If the type is user, then the "dest" value is the user's extension number. If the type is group, then the dest value is the name of the group. If the type is station, then the dest value is the station number.

A trunk group for lines that are on a PRI or are SIP trunks can also specify the outgoing caller ID that will be used when outgoing calls are made on these lines. To specify the outgoing caller ID number:

<trunk_group=0><outgoing_cid_number=9729859901>

The phone company may or may not allow you to specify the outgoing caller ID number. If they allow it, they will probably require that the outgoing caller ID number be one of the DIDs that you have on your service. You may also be able to specify an outgoing caller ID name. If the phone company allows this, you specify the outgoing caller ID name like this:

<trunk_group=0><outgoing_cid_name=FacetCorp>

Defining Speed Dial Numbers in sys_speed_dial.cfg

The sys_speed_dial.cfg file is where speed dial numbers are defined. You can have as many of these as the range defined in sys.local.cfg for speed_dial_low and speed_dial_high will allow. After changing this file, from a shell prompt logged in as root run the command:

     facetphone reload

Each entry looks like:

<speed_dial=20><number=9,972-555-1111><desc=Acme Sales>

The value in the speed_dial item is the number that users will dial to call the number associated with the speed dial.

The value in the number item is the number to be dialed including the leading 9 or 8 plus trunk group number. The comma and dashes used in the example are not necessary but help make the number easier to read.

The value for the desc item is a description of who the speed dial calls. This can be up to 100 characters long.

Defining Call Routing Rules in call_routing.cfg

The call_routing.cfg file defines call routing rules. These rules can be used to route calls to the trunk group where the cost of the call will be the least. They can also be used to add or remove parts of the dialed number for proper dialing on the trunk group being used. Each entry looks like:

<match=1214><dial=81[1214]+><optional=y>

The value of the "match" item is a dialing pattern to be matched to the beginning of a number the user dialed using the default trunk group (usually by dialing 9 unless this has been redefined). Note that the leading 9 to select the default trunk group is not used as part of the number to match – only the part after the 9.

The value of the "dial" item is a dialing editing pattern to be used to create the new number to be dialed including the specific trunk group specification (by dialing 8 plus the trunk group number unless this has been redefined). In the new pattern to be dialed, brackets indicate digits matched in the number the user dialed which should be removed. A plus sign indicates to copy the remainder of the digits that the user dialed to the new number to be dialed. In the example above, the leading 1214 that the user dialed will be removed

The third part of the entry indicates whether the new routing is optional or not. If the designated trunk group has all its lines busy, and the third part of the entry is <optional=y>, then the number will be dialed on the default trunk group just as the user dialed it. If the third part of the entry is <optional=n> then the new routing is forced and the user will receive a message that all the lines are currently in use.

Let's go through the use of the example routing rule above:

- A user dials 912145551212

- Because he dialed 9 for the default trunk group, the call routing entries will be evaluated.

- Discarding the leading 9, the number dialed matches the pattern 1214 so this call routing rule will be used.

- The new number to be dialed will start with 81, choosing trunk group 1 for making the outgoing call.

- The 1214 part of the number is removed according to the rule.

- The remainder of the number is added to the new number.

- The resultant number to dial is 815551212, using trunk group 1 to dial the local number 555-1212.

Another form of a call routing rule begins with a trunk group specification to indicate that the rule is only to be used if the default outgoing trunk group for the call be dialed is on the specified trunk group. In the following example, trunk group 2 has lines that are SIP trunks which require all calls to be dialed with a leading 1.

```
<trunk_group=2><match=1><dial=9+><optional=y>
<trunk_group=2><match=><dial=91+><optional=y>
```

These two entries are for use only when the user is making a call that will go out on a line in trunk group 2 by default. In this case, these rules are not being used to direct the call out a different trunk group, but instead to just make sure there is a leading 1 and then sent out the same default trunk group. Because the rules are evaluated in the order that they appear in the file, the first rule will pass a number that begins with one on through by just putting the leading 9 back in and using the number as dialed. If the number did not begin with a 1, then the second rule is evaluated and it adds the leading 9 and 1 before copying in the rest of the number as dialed.

After changing the call_routing.cfg file, from a shell prompt logged in as root run the command:

    facetphone reload

Configuring Names of IVR Scripts and Prompts in ivr.local.cfg

The ivr.cfg and ivr.local.cfg files are used to specify the names of IVR scripts that can be chosen as auto-attendants, voice mail handlers, on-hold scripts, etc. They also include the name of prompts that can be choosen from a drop list for recording. The ivr.cfg file has standard entries and is overwritten each time the system is upgraded. Any local changes should be made in the ivr.local.cfg file. Entries in the file look like this:

```
<ivr_script=acme_open>
<ivr_prompt=acme_open_greeting>
```

The value for an ivr_script item defines a name that is available for selecting as an IVR script.

The value for an ivr_prompt item defines a name that is available for recording a prompt.

You can also remove names that are in the ivr.cfg file that you do not want to appear on your system. To do this, add <remove> to the beginning of the line:

```
<remove><ivr_script=company_open_menu>
<remove><ivr_prompt=company_open_greeting>
```

The above entries would remove the company_open_menu script defined in ivr.cfg from the list of scripts available to choose from and remove the company_open_greeting from the list of prompts that can be recorded.

After changing this file, from a shell prompt logged in as root run the command:

	facetphone reload

Defining Skills in skills.cfg

The skills.cfg file contains the definition of skills that will be used to do skills based ACD call routing. If you are not going to do this, then nothing needs to be put into the skills.cfg file. Each entry in the file looks like:

<skill_id=0><name=Sales>

The skill_id number should be unique for each skill entry. When entering a list of skills for a user, they will be validated against the list of skills in the skills.cfg file.

After changing this file, from a shell prompt logged in as root run the command:

facetphone reload

Defining Groups With the FacetPhone Administrator

Groups are used in several different ways in the FacetPhone configuration:

- Groups can be used to define the people watched in a Users window and the calls watched in a Calls window. The members of the group will be in the Users window and any calls they are a party to will be in the Calls window for that group.

- A group can be used to provide a group voice mailbox that can be accessed by multiple people.

- A group is used as the context for an auto-attendant. Lines are assigned an operator group and when a call comes in on the line, the auto-attendant settings for the operator group are used to determine how the call is handled. Calls coming in on lines may be handled by a different group if the call has a DID that is directed to a different group.

- The operators defined for a group can be added to a call allowing backup operator groups to be defined and used if the operators in the primary group are all busy or unavailable.

- A group can be used to define a zone of users to send pages to.

- Finally a group is used to define the users who will be available in the company directory where callers can find users by name.

To define a new group, on the FacetPhone Administrator menu choose:

Add -> Add a Group

If you are modifying an existing group choose:

Configure -> Configure Groups

If you are configuring an existing group, you will choose it from a list:
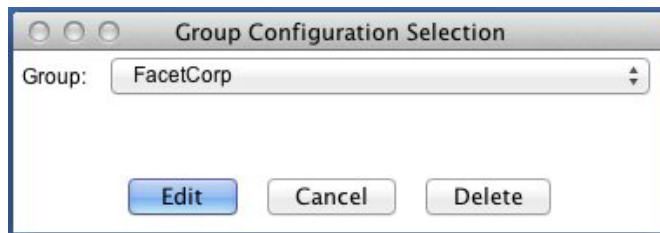
Select the group from the list and click on the Edit button to modify it. Note that you can delete the group from this screen too.

When you add a new group or change an existing one, the Group window is opened:

Group Facet Corp

Group name: Facet Corp

☐ All users   [Members]   [Operator Stations]

☑ Anyone can answer any ringing call   ☑ Managers can monitor and barge in

Ring for new incoming calls: 0

Ring for transferred calls: 0

Ring back seconds for call on hold/parked: 300

Home branch: Dallas office ▲▼

Auto–attendant settings

☑ Auto-attendant ON

ON script name: facetcorp_open_menu ▲▼

Overflow script name: company_overflow_menu ▲▼

Rings before overflow: 4

ON HOLD script name: music_on_hold_ring_back ▲▼

Automatic Call Distribution (ACD): Off ▲▼

Wrap-up seconds for ACD oper: 0

Queue priority: 0

Group display privacy settings

User list privacy level: Anyone can watch ▲▼

Call list privacy level: Anyone can watch ▲▼

Mailbox privacy level: Anyone can watch ▲▼

Station list privacy level: Anyone can watch ▲▼

Parked/queued calls privacy level: Anyone can watch ▲▼

Lines list privacy level: Anyone can watch ▲▼

[Voice Mail Handling]

Call recording

☐ Record all incoming calls to mailbox

Recordings to mailbox: Facet Corp ▲▼

[OK]   [Cancel]

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The settings on this screen are:

Group name

The group name should be descriptive and not contain spaces or any characters not allowed in a file name.

All users

Checking this box will make every user on the system automatically a member of this group.

Members

When you click on the Members button, the screen used to define the members, operators and managers of the group will be displayed:
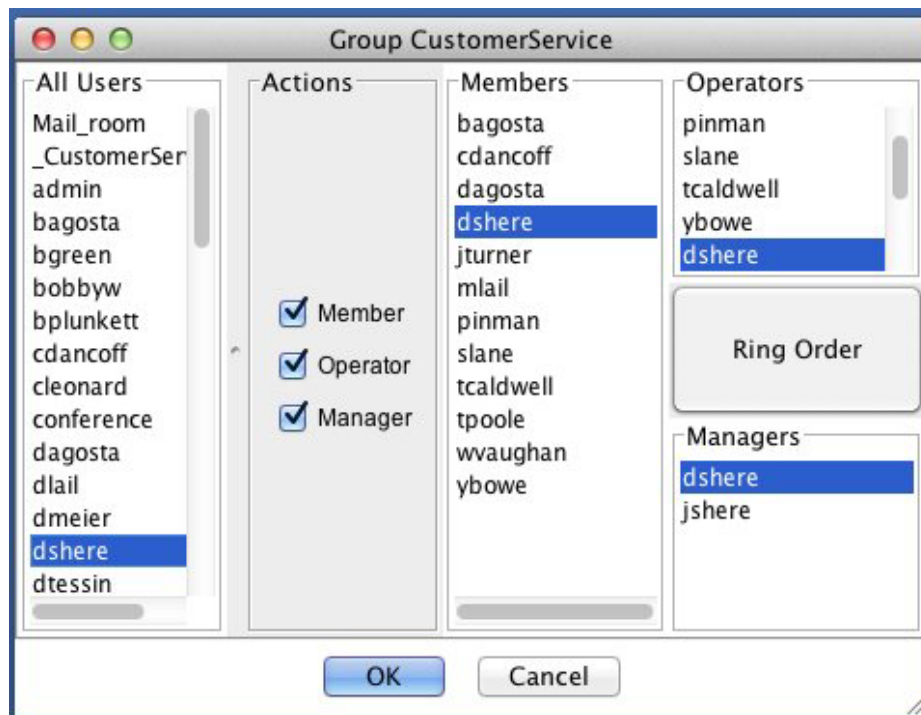
The panel on the far left has a list of all users defined on the system. When you click on a user in this list, then the "Member", "Operator" and "Manager" check boxes will show these attributes for the selected user. You can check or uncheck the boxes to get this right for the selected user.

The order of the operators does not matter except in the case where fixed order ACD is being used. If this ACD method is being used, then click on the "Ring Order" button and a window will be presented where you can change the order:
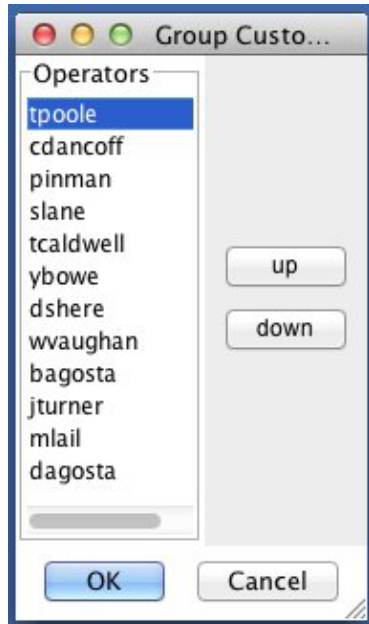
Click on an operator and then use the up and down buttons to move their location in the list. In fixed order ACD, the operators will be checked for availability to take an incoming call from the top of this list to the bottom.

Operator Stations

Operator stations are rung on an incoming call to the group regardless of being associated with a user or their status. If an ACD method is being used, an operator station will be rung on incoming calls but will not be used to determine the availability of an operator in deciding whether to queue the call or not. To define operator stations for the group, click on the "Operator Stations" button and the operator station window will be displayed:
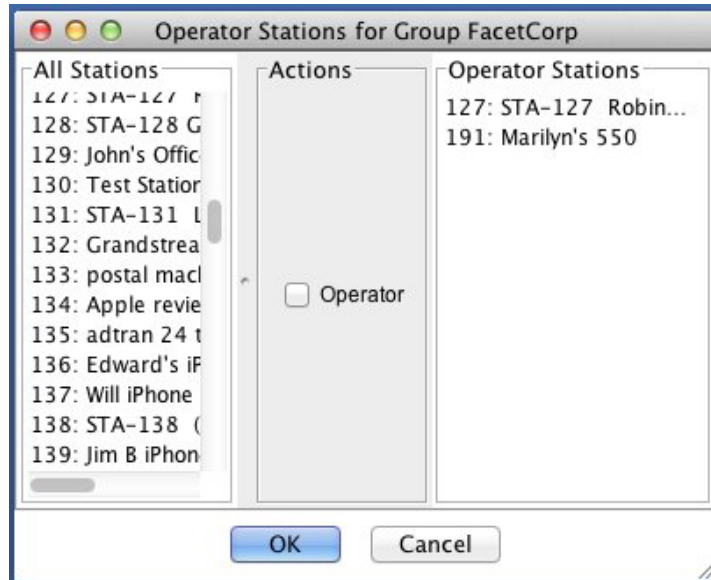
The left panel shows all the stations defined on the system. When you click on one, then you can use the "Operator" checkbox to determine whether that phone will be an operator station for the group.

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 window 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.

Ring for new incoming calls

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 station-to-station calls 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 station-to-station calls. A distinctive ring value of -1 for a group or the inside ring type will cause the phone to use the ring type that it has been locally 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 or on hold 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.

Auto-attendant ON

> If this box is checked, then the auto-attendant will be invoked immediately for an incoming call to this group.

ON script name

> This specifies the auto-attendant script to use when the auto-attendant is turned on. Usually you will create scheduled events for turning the auto-attendant on and off and selecting the script. However, if it is never changed, you can set it here.

Overflow script name

> This is the script that is invoked after some number of rings when the auto-attendant is off. This script is also invoked when the auto-attendant is off and all the operators are busy and do not have call waiting turned on.

Rings before overflow

> 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 is the script to be run when a call in this group is put on hold or parked. The standard scripts are:
>
> - music_on_hold – This script plays songs. The standard music provided with the system consists of 10 royalty free instrumental songs.
>
> - message_on_hold – This script plays a message repeatedly. The prompt to record for this message is "on_hold_message".

Automatic Call Distribution (ACD)

> This setting determines whether automatic call distribution is turned on for the group and, if so, which method is used. The values for this setting are:

- Off – When ACD is off, calls to the operators in this group will ring to all available operators and operator stations at the same time.

- Round robin – This ACD method selects the next operator to ring based on which one took the last call for the group. It starts searching the for the next operator from the last operator's position in the list of operators.

- Longest idle – This ACD method selects the next operator based on which one has not been in a call answered as an operator for the longest time.

- Fixed order – This ACD method always searches for an available operator in the same order starting with the first operator in the list.

Note that if ACD is turned on, you MUST have an auto-attendant on to control it.

Wrap-up seconds for ACD oper

If ACD is turned on, this is the amount of time after an operator hangs up in an incoming call before another ACD call is sent to them. This allows the operator some time to perform any tasks needed for the call that just ended before having to take a new call.

Queue priority

This is the priority for the group's queue. If operators are in multiple groups and calls are queued to multiple groups, then when one of these operators becomes available, the system will choose the next call from the queue of the group with the highest priority. The highest priority is 1, the next highest is 2, etc. If a group's priority is blank or 0, then it has no priority and no calls will be taken from its queue until all queues with a specified priority have been emptied.

Privacy levels

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, lines and voicemail. 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.

Voice Mail Handling

When you click on the "Voice Mail Handling" button, the voice mail configuration screen for the group will be displayed:
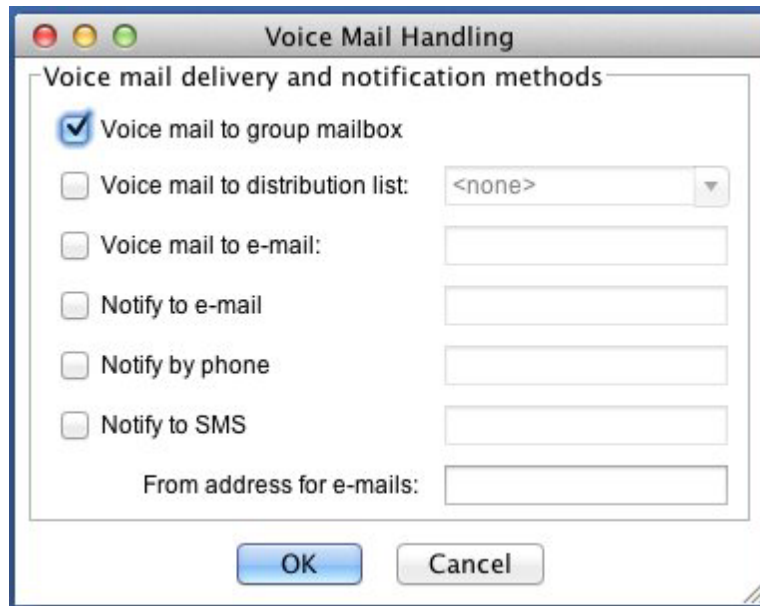
You can choose one or more delivery methods for voice mails that are left in the group's mailbox. The options are:

- Voice mail to group mailbox – If this box is checked then the voice mail will be in the mailbox that is accessible from the user interface and the user menu on the phone. It is almost always best to have this box checked.

- Voice mail to distribution list – If this box is checked, then choose the voice mail distribution list from the drop list. This will cause voice mails for this group to be sent to the individual voice mail boxes for each of the people in the distribution list.

- Voice mail to email – If this box is checked, then enter the email address where the voice mail should be sent. The voice mail will be sent as a .wav file attachment to the email. Multiple email addresses can be specified and must be separated by semi-colons or commas.

- Notify to email – If this box is checked, then enter the email address where the notification email should be sent. Note that this will not cause the voice mail to be sent as an attachment. Only the caller ID information related to the voice mail will be included in the email. Multiple email addresses can be specified and must be separated by semi-colons or commas.

- Notify by phone – If this box is checked, then enter the phone number that you want called with the notification. You should enter a leading 8 and the trunk group that you want used before the phone number. To call a list of numbers, separate the numbers with a comma. When a voice mail for the group arrives, the system will call the number(s). The person receiving the call will be prompted to press * and then their user ID and PIN. They will then be in the user menu and can listen to the voice mail. The system will continue making this call until it gets the * response from one of the people called.

- Notify to SMS – If this box is checked, then 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 semi-colons or commas.

- From address for emails – If you have chosen any option which will send an email, then you must enter the email address that you want used in the From header of the email being sent.

Call Recording

If you check the box "Record all incoming calls to mailbox", then it will record all incoming outside calls coming to this group. Unlike turning on record all calls for a user, these

recordings will be associated with the line the call came in on and will provide a continuous recording of the caller and all interactions with inside parties and auto-attendants. If the call is forwarded back out to another outside number, the recording will continue as well.

If this feature is turned on, you can select the group voice mailbox to be used for the recordings from the drop list.

Defining Gateways With the FacetPhone Administrator

A gateway is a physical or virtual device that connects phone service, analog phones or other analog devices such as paging systems to the FacetPhone system. Analog lines from the phone company plug into an FXO gateway. Analog phones connect to an FXS gateway. Paging systems may connect to either depending on the model. These devices make the conversion from analog voice to digital voice over IP. We also need to define a gateway to connect to SIP trunks even though there is no physical gateway involved. The settings for the SIP trunk "gateway" describe how to connect to the SIP trunk service.

To define a new gateway, on the FacetPhone Administrator menu choose:

> Add -> Add a Gateway

If you are modifying an existing gateway choose:

> Configure -> Configure Gateways

If you are configuring an existing gateway, you will choose it from a list:
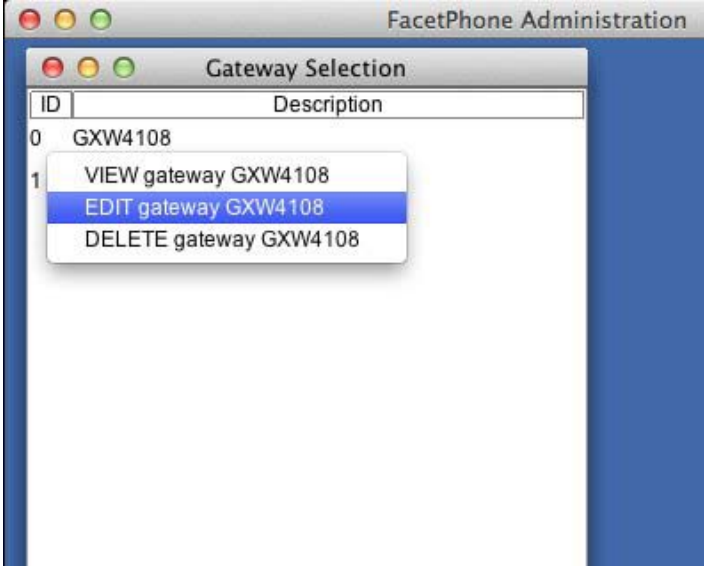
Whether you are adding a new gateway or have selected an existing gateway to edit, then the Gateway Configuration screen will be presented:

Gateway Grandstream GXW4104

Gateway ID:	3
Name:	199.29.166.164
Gateway description:	Grandstream GXW4104
IP address of the gateway:	199.29.166.164
MAC address of the gateway:	000b825f977a
Default route:	199.29.166.78
Manufacturer:	Grandstream GXW410x
Number of FXS ports:	0
Number FXO ports or T1/SIP Channels:	4
Branch where gateway located:	Dallas office

☐ Digits detected in audio   ☑ RFC2833 digit detection

SIP trunking service settings

Gateway ID to consolidate lines:	
URI Domain:	
Registrar:	
Number:	
ITSP Domain:	
Username:	
Password:	
Expires:	60
SIP socket number:	1
NAT Address:	

☐ Allow multiple SIP servers to send incoming calls to the same number

☐ Use RFC 2617 quality of protection (qop)

☐ Provider sends Ring before Session Progress

☐ Use E164 phone number format

[OK]   [Cancel]

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The following settings are used for gateways of all types.

Gateway ID

> Each gateway has a unique numeric ID that is referenced by the lines or phones that are attached to it. Gateway IDs begin with 0 and you can have as many as configured by the max_gateways setting in sys.local.cfg.

Name

> The gateway name is generated by the system and is here for information only.

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. If there are multiple branches, it is also useful for the description to indicate where the gateway is.

IP address of the gateway

> Enter the IP address that the gateway has been configured with. All media gateways on the FacetPhone system should have fixed IP addresses.

MAC address of the gateway

> Enter the MAC address of the gateway. For most media gateways this is not necessary, but if you have it you might as well enter it.

Default route

> The default route is the IP address of the network router that can route packets if the FacetPhone server and the gateway are not on the same subnet. If the gateway is on the same subnet with the server, this field may be left blank.

Manufacturer

> Choose the gateway manufacturer (and possibly model) from the list.

Number of FXS ports

>If this gateway has FXS ports that connect to analog phones, then enter the number of FXS ports that the gateway has.

Number of FXO ports or T1 channels

>If the gateway has FXO ports that connect to analog phone lines, then enter the number of FXO ports here. If the gateway connects to a PRI, then enter the number of channels that are active. A full T1 configured as a PRI has 23 channels.

Branch where gateway located

>Choose the branch where this gateway is located from the list.

Digits detected in audio

>Most modern VoIP media gateways now use RFC2833 to detect digits during a call. However, if the gateway does not support it, then check the box for detecting the digits in the audio stream.

RFC2833 digit detection

>This is the mode of digit detection during a call that you will use unless you know that the gateway does not support it.

The following settings are only used when the "gateway" is a description of a SIP trunking service.

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.

SIP socket number

FacetPhone supports up to 3 SIP socket addresses for communicating with gateways or SIP phones. These are specified in sys.local.cfg. This setting should have a value of 1, 2 or 3 to specify which SIP socket number has the proper SIP port specified for communicating with this gateway.

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.

Allow multiple SIP servers to send incoming calls to the same number

Check this box if the SIP service provider sends calls to the same DID from multiple addresses.

Use RFC 2617 quality of protection (qop)

Check this box if the SIP service provider requires implementation of RFC 2617 in registration authentication.

Provider sends Ring before Session Progress

Check this box if the SIP service provider sends a Ringing response to an outgoing INVITE before sending the Session Progress response. Most service providers will only send the Session Progress response which allows the connection to be established and the ring tone played by the provider.

Use E164 phone number format

This is a format required by some SIP trunk providers that specifies that all phone numbers start with "+1". Note that this is only between FacetPhone and the SIP provider. Users can still dial local numbers and FacetPhone will adjust it for sending to the SIP provider.

Defining Lines with the FacetPhone Administrator

A line is a physical or virtual circuit from your phone service provider. All lines are defined as being connected to a port on a gateway. In the case of an analog line, it actually is plugged into a port on an FXO gateway. In the case of a PRI, the channels are assigned to ports 1 to 23 (or less if a fractional T1 is being used). If the service is provided as SIP trunks, then each of these virtual lines are specified as being connected to their own unique port on the gateway.

To define a new line, on the FacetPhone Administrator menu choose:

> Add -> Add a Line

If you are modifying an existing line choose:

> Configure -> Configure Lines

If you are configuring an existing line, you will choose it from a list:
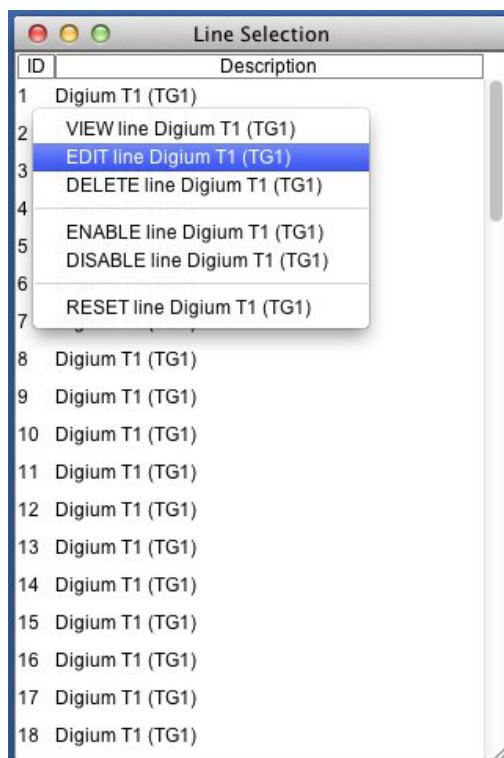
Whether you are adding a new line or have selected an existing line to edit, then the Line Configuration screen will be presented:

The settings for a line are:

Line ID

> This is a unique number used to refer to this line. If you are adding a new line, there will also be a checkbox that you can check to use the next available line number for the new line.

Gateway

> Choose the gateway this line will connect to from the drop list of gateways that have been defined.

Gateway port

> Choose the port that this line will connect to on the gateway. In the case of an FXO gateway for analog lines, this will be the physical port the line plugs into. In the case of PRI or SIP trunk channels, this port number just needs to be unique for this line. Port numbers begin with 1 and increase to the number of ports on the gateway.

Line description

> You can enter any description you want here, but it is best to include information such as the phone number for the line and possibly the gateway and port it is connected to. You might also want to include the trunk group that the line belongs to.

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.

Connected to paging system

If this line is on an FXO gateway port and 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 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. This setting is usually only used on a line temporarily for diagnostic purposes.

Route audio through 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. This is only needed if the gateway is not able to have its voice packets routed directly to the endpoint at the other branch.

Route audio through 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. Some gateways are not able to properly handle this sort of "hairpin" call.

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.

Dial 9 for Centrex outside line

This is a legacy feature of analog phone lines in which a Centrex service requires that a phone connected directly to the line must dial 9 before getting an outside dial tone. This is rarely used now, but if your lines have this kind of service, 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, enter those digits in the "Centrex transfer suffix".

Centrex transfer prefix

If Use Centrex transfer is checked and it is necessary to dial digits before performing the flash, enter those digits here.

Centrex transfer suffix

If Use Centrex transfer is checked and it is necessary to dial digits after the flash, enter those digits here.

Line gives caller ID

If the line delivers caller ID, check the box for that feature. All digital lines have this feature. On analog lines it may be an optional service. If the line does not have caller ID, then having this box unchecked will cause the system to answer incoming calls earlier instead of waiting for the caller ID to arrive between the first and second rings.

Defining IP Phones With the FacetPhone Administrator

An IP phone is a telephone instrument that plugs into your local area network instead of an analog phone line. It may also be a "soft" phone that is a program running on a computer or an app running on a smart phone.

To define a new IP phone, on the FacetPhone Administrator menu choose:

>   Add -> Add an IP Phone

If you are modifying an existing line choose:

>   Configure -> Configure IP Phones

If you are configuring an existing phone, you will choose it from a list:
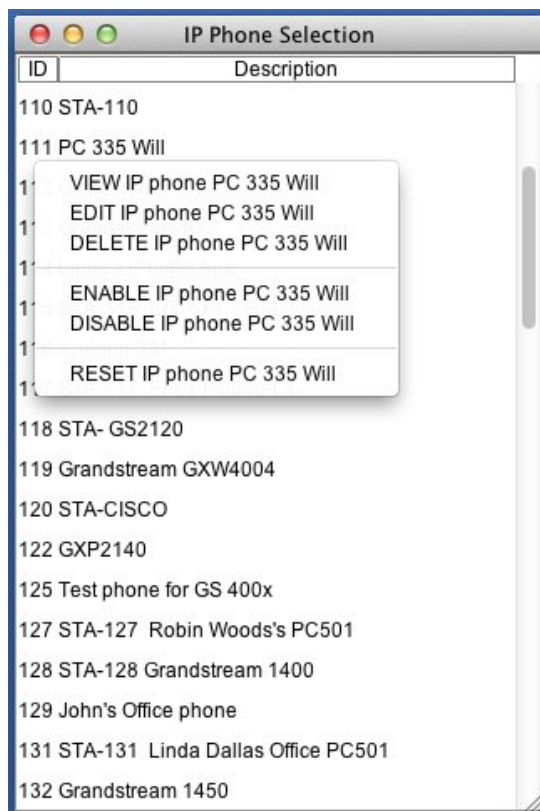
Whether you are adding a new IP phone or have selected an existing phone to edit, then the IP Phone Configuration screen will be presented:

The settings for an IP phone are:

Station Number

>Enter the station number for the IP phone. If you are defining a new phone, there will be a check box to use the next available station number if you want to choose it that way.

Manufacturer

>Choose the IP phone manufacturer and model from the list.

IP phone description

>Enter a description for the phone. We suggest you include information about the location of the phone.

Branch where IP phone located

>Choose the branch where the phone is located from the drop list. Branches are defined in the branch.cfg file.

Operator group

>Choose the operator group that this phone will belong to. This gives calls from this phone a group association when no user is associated with the phone. It also determines which phones are displayed in a group's Stations window on the user interface.

Dialing restriction

>You can choose a dialing restriction from the drop list to be used when no user is associated with the phone. When a user is associated with the phone, the user's dialing restriction, if any, will apply instead of the phone's dialing restriction.

Default outgoing trunk group(s)

>Choose the default outgoing trunk group(s) for the phone. This determines which lines will be used when a user makes an outside call from this phone by dialing 9 before the outside number. 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.

Share license with station

> This setting is only available when the manufacturer chosen is "FacetPhone mobile app". It allows you to specify another IP phone to share its license with the mobile app. When the app logs in, the other IP phone will be disabled. When the app logs out, the other IP phone will be enabled. If you want to use this feature, enter the station number of the other IP phone that will share its license with the mobile app.

Dynamic address

> If this box is checked, then the phone is assumed to have a dynamic IP address provided by a DHCP server. If the phone is outside of the LAN, then this box should be checked regardless of how the phone actually gets its address. This lets the system know that it will learn the phone's address when it registers.

IP address of the phone

> If the phone has a fixed IP address, enter its address here.

MAC address of the phone

> Enter the MAC address of the phone. This can be found on a sticker somewhere on the phone. It is a 12 digit hexadecimal number.

Default route

> The default route is the IP address of the network router that can route packets if the FacetPhone server and the IP phone are not on the same subnet. If the IP phone is on the same subnet with the server, this field may be left blank.

NAT address

> If the IP phone is not on the LAN with the server, then enter the public IP address that packets from the phone will appear to come from.

Domain

This is the domain name that will be used in authentication of the phone.

User name

This is the user name that the phone will use when authenticating. If the IP phone is a Polycom or Grandstream, then this will be put in the configuration file generated for the phone. Otherwise, you must manually enter the same user name on the phone as you have entered here.

Password

This is the password that the phone will use when authenticating. If the IP phone is a Polycom or Grandstream, then this will be put in the configuration file generated for the phone. Otherwise, you must manually enter the same password on the phone as you have entered here.

Expires

This setting tells the phone how often to register. It only has an effect on Polycom and Grandstream IP phones where the system generates the configuration file for the phone.

SIP socket number

FacetPhone supports up to 3 SIP socket addresses for communicating with gateways or SIP phones. These are specified in sys.local.cfg. This setting should have a value of 1, 2 or 3 to specify which SIP socket number has the proper SIP port specified for communicating with this gateway.

Digits transmitted in audio

This setting indicates that digits pressed during a call with the phone will be sent in the audio stream. This is usually not set.

Transmit digits as RFC 2833 packets

This setting indicates that digits pressed during a call with the phone will be sent using packets as described in RFC 2833. This is the normal setting for most IP phones.

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. This setting is usually necessary for phones that are outside of the LAN unless they are on a VPN.

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. This is only needed if the gateway is not able to have its voice packets routed directly to the endpoint at the other branch.

Defining Analog Phones Connected to a Gateway

Although most phones used on FacetPhone systems are IP phones, it is possible to use analog phones. These are connected to an FXS gateway which must be defined first. Sometimes paging systems are also connected to an FXS port and they are also defined as an analog phone connected to a gateway.

To define a new analog phone, on the FacetPhone Administrator menu choose:
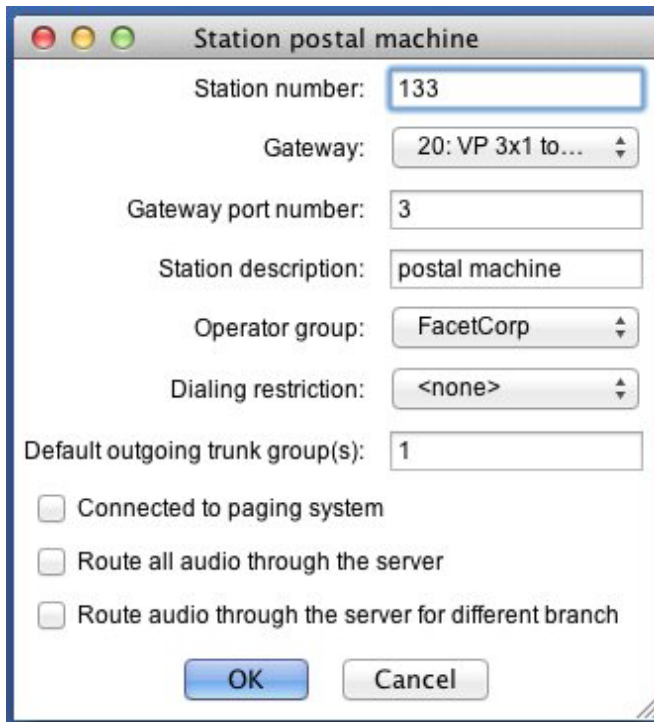
Add -> Add an analog phone (on a gateway)

If you are modifying an existing analog phone choose:

Configure -> Configure analog phones (on gateways)

If you are configuring an existing phone, you will choose it from a list:
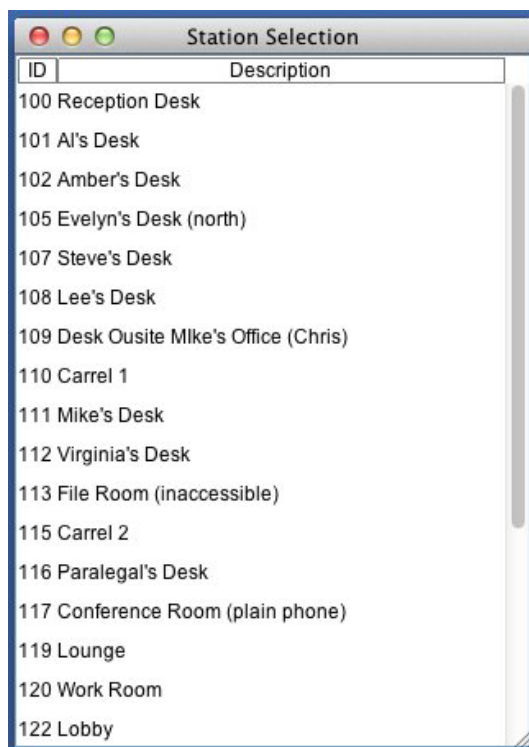
Whether you are adding a new analog phone or have selected an existing phone to edit, then the Line Configuration screen will be presented:

The settings for an analog phone on a gateway are:

Station number

> Enter the station number for the analog phone. If you are defining a new phone, there will be a check box to use the next available station number if you want to choose it that way

Gateway

> Choose the gateway this phone will connect to from the drop list of gateways that have been defined.

Gateway port number

> Choose the port that this phone will connect to on the FXS gateway. Ports are numbered starting with 1 and increasing to the number of ports on the gateway.

Station description

> Enter a description for the phone. We suggest you include information about the location of the phone.

Operator group

> Choose the operator group that this phone will belong to. This gives calls from this phone a group association when no user is associated with the phone. It also determines which phones are displayed in a group's Stations window on the user interface.

Dialing restriction

> You can choose a dialing restriction from the drop list to be used when no user is associated with the phone. When a user is associated with the phone, the user's dialing restriction, if any, will apply instead of the phone's dialing restriction.

Default outgoing trunk group(s)

> Choose the default outgoing trunk group(s) for the phone. This determines which lines will be used when a user makes an outside call from this phone by dialing 9 before the outside number. 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.

Connected to a paging system

> If this analog phone is actually a paging system connected to an FXS port, then check this box.

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. This setting is usually not needed for analog phones on a gateway except for diagnostic purposes.

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. This is only needed if the gateway is not able to have its voice packets routed directly to the endpoint at the other branch.

Defining Users with the FacetPhone Administrator

There are many details defined for each user including the complete description of all the user interface displays and their location on 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

This will cause the User Selection screen to be presented:
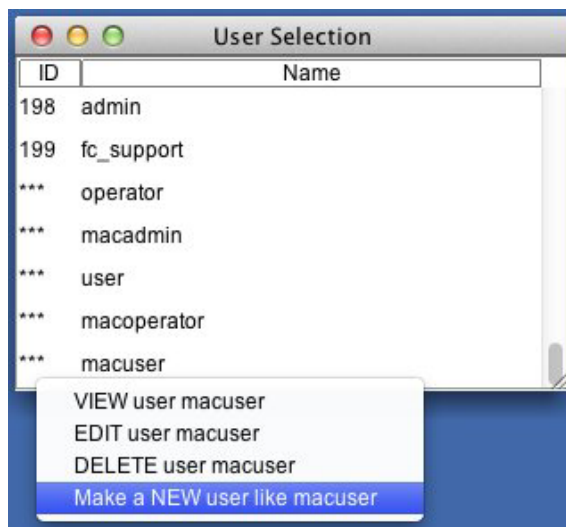
To create a new user, click on the user and choose the "Make NEW user …" option to create a new user based on the user or model you have chosen. The models provided with the system are:

- user – This is for a normal user with the windows laid out for a Windows PC.

- macuser – This is for a normal user with the windows laid out for a Mac.

- operator – This is for a user who will be an operator on the system with the windows laid out for a Windows PC.

- macoperator – This is for a user who will be an operator on the system with the windows laid out for a Mac.

You can also choose a real user to copy for a new user.

When you create a new user, the first window that is presented is one where you enter the new user's extension or have the system choose the next available extension. You also enter the user's login name. If you are creating a new model based on one of your users, then check the box for "Model Only". Models do not have extension numbers.

When you click OK to this window, then the main user edit window will be displayed. This is also the window that is displayed when you are editing the settings for an existing user:

We will cover the settings on this main screen first and then proceed to the Locations screen and the Voice Mail Handling screen. The main user's screen as viewed from the administrator function has more settings than when the user modifies his settings himself. For

example, the privilege and restriction settings are not available for the user to change but only the administrator.

The first settings on this screen provide general details about the user:

User extension

> This is the extension number for the user. It is the number that other users will dial to reach the user and that outside callers will dial when they are allowed to dial an extension. Remember that a user's extension number is distinct from a phone's station number. Users can be reached at their extension number regardless of which station they are currently using.

User name

> This is the login name that the user will use when logging in to the FacetPhone system with the user interface.

First name

> This is the user's first name. It will be searchable by callers in the company directory.

Last name

> This is the user's last name. It will be searchable by callers in the company directory.

PIN

> This is the PIN that the user will enter when logging in to FacetPhone with the user interface or logging in on a phone with the user menu.

Home branch

> This is the branch where the user usually works. This is used to assign a time zone to the user's voice mail times.

Skills

> This is a comma separated list of skills that this user has. Each skill must match one defined in skills.cfg.

The Privileges section has settings that can be used to grant certain privileges to the user:

Admin

> If this box is checked, then this user has administrative privileges. This will cause some settings to be disabled in the restrictions section.

Allow mobile app usage

> This box must be checked in order for the user to connect to the phone system with the FacetPhone mobile app on a smartphone.

Hide outside calls

> If this box is checked, then the user's outside calls will not be visible in the Calls windows on other user's computers.

The Restrictions section has settings that can be used to restrict what the user can do:

Deny configuration

> If this box is checked, then the user is not allowed to make changes to their configuration.

Deny call recording

> If this box is checked, then the user cannot initiate or stop recording of calls. If you want a user to have all their calls recorded and they cannot change this, then check the boxes for "Record all calls to mailbox", "Deny configuration" and "Deny call recording".

Deny forward v-mail to email

> If this box is checked, the user will not be able to forward any voice mail to an email address.

Deny conferencing

> If this box is checked, the user will not be able to create ad hoc conferences. They can still call a conference bridge extension and be part of a conference in the conference bridge.

Must dial job code

> If this box is checked, the user will be required to dial a job code on their phone before an outgoing call will be completed.

Call display on answer

> If this box is checked, then as soon as a user answers an incoming outside call, the call display window will be popped up where they can enter a job code or note about the call. This feature does not apply to incoming calls from another extension.

Dialing restriction

> You can choose a dialing restriction from the drop list to be used on whatever phone is currently associated with this user. If the phone has a dialing restriction, the user's dialing restriction will override it when the phone is associated with the user.

Pre-call script

> This setting allows you to define a script to be run before the user places an outside call. For example, you might want to run a script to be sure that the user is not dialing a number on a do-not-call list.

The next group of settings determine how call recording is handled for the user:

Record all calls to mailbox

This setting determines whether or not all of the user's calls will be automatically recorded.

Recordings to mailbox

This setting determines where the user's call recordings will be stored. By default, they are stored in their personal voice mail folder but a group voice mail folder may also be chosen.

The last group of settings allow you to specify outgoing caller ID for the user:

Outgoing CID number:

This setting is only used when the user is making an outgoing call on a PRI or SIP trunk where the system can provide an outgoing caller ID. Usually the number must be one of the DIDs provided by your phone service. For example, if the user has a DID that rings outside calls directly to him, then it might make sense to use that number for the user's outgoing caller ID number.

Outgoing CID name:

This setting is only used when the user is making an outgoing call on a PRI or SIP trunk where the system can provide an outgoing caller ID. Even in this case, many service providers will not use the outgoing caller ID name provided by the phone system. You can enter any alphanumeric name you want here that is up to 15 characters long.

To access the user's location profiles, click the "Locations" button. To review from the User's Guide, location profiles define what phone a person will use at each location. In addition there are other settings that are associated with the profile to customize the settings for each particular location.

Note that the user's and the administrator's view of location settings are the same. Therefore the same information is in the User's Guide but is repeated here for convenience.

When you click the Locations button the Location Selection screen will be displayed:

There are three pre-defined locations – "office", "not logged in" and "FP Mobile app".

The "office" location profile is meant for defining the settings for the user at their main work location.

The "not logged in" location profile is used to describe options in effect when the user is not logged in to the user interface. If you have their office phone associated with this profile, then if somebody calls them when they are not logged in, that phone will still ring. If they do not have a phone associated with the not logged in profile, then when users call them and they are not logged in, calls will go directly to their voice mail.

The "FP Mobile app" location profile is used if the user has the FacetPhone mobile app installed on a smart phone and they are logging in with the app. This location profile will have the station number that has been assigned to the user's mobile app.

When you edit an existing location profile or add a new one, the following window is presented:

Location name

> This is the name of the location. It is not editable.

Voice type

    The voice type is set to either "Station" or "None". You will almost always want to choose "Station" unless this is the user's "Not logged in" location and they want their calls to go directly to voice mail when they are not logged in.

Station number

    Set this to the station number of the phone the person will use at this location.

Flash treated like hangup and pickup

    This setting is only used if the person has an analog phone at this location. They purpose is to prevent hanging up on a call and then picking the receiver up quickly to make another call and inadvertently putting the first call on hold instead of hanging up on it. If you check this box, the user's analog phone will not be able to put a call on hold with the flash button.

Dialtone after call ends

    This setting is only for IP or analog phones on a gateway which use the MGCP protocol. This setting allows the user to stay off-hook after a call and a new dial tone will be presented so that they can immediately dial again.

Auto offhook on intercom calls

    This setting determines whether the user's phone will automatically answer a call from another extension. When the phone does this, it will give a short beep to let the user know it has gone off-hook and is in a call. This only applies to calls from another extension. Any calls from outside the company will always ring the phone.

Page to phone

    This setting determines whether pages made on the phone system are played on the user's phone.

Sounds during instant messaging

This setting determines whether sounds are played on the user's computer speakers when an instant messaging session is started and when messages are sent and received.

Play pages on PC speaker

This setting determines if and when pages made on the phone system are played on the user's computer speakers. You can choose "Never", "Always" and "When not on the phone".

Speaker volume

This setting allows you to adjust the volume of sounds, voice mail and pages played over the computer speakers. The lowest numbers are quietest and the higher numbers are loudest.

The settings grouped together as "Incoming call when your are not on another call" determine what happens when the user's phone is idle and a call rings to you. You must have at least one of these options selected.

Ring phone

If this box is checked, then new calls coming to the user when their phone is idle will ring their phone.

New call notify box

If this box is checked, then new calls coming to the user when their phone is idle will cause the call notification window to be popped up where they can answer the call or send it to voice mail.

The settings grouped together as "Incoming call when you are on another call" determine whether the user has call waiting enabled. If they do not want call waiting and want calls to them while they are on the phone to go to voice mail, then do not check either of these boxes. If call waiting is enabled, when they answer a call waiting their current call will be put on hold.

Call waiting tone

> If this box is checked, then new calls coming to the user when they are already on a call will cause the call waiting tone to be played on their phone.

Call waiting notify box

> If this box is checked, then new calls coming to the user when they are already on a call will cause the call notification window to be popped up where they can answer the call or send it to voice mail.

Notify box ring sound

> This setting allows you to select a sound to play on the user's computer speakers when a call notification window is presented. As you select each one it will play once so that you know what it sounds like. If you choose "None", then no sound will be played with a call notification window.

The settings grouped together as "Voice mail" allow you to configure a voice mail notification window to be popped up when the user has a new voice mail.

Notify box for new voice mail

> Check this box if the user wants a voice mail notification window popped up when they receives a new voice mail.

New voice mail notification sound

> If the user wants a sound associated with the voice mail notification window, you can choose it here.

Minutes between VM sounds

> The voice mail notification window sound will repeat as often as you specify here. If the setting is 0, then it will only be played once and not repeated.

Default status

>This setting is where you choose what the user's status will be as soon as they login at this location. If you set the default status to "<none>", then the user's status will not be changed when they login at this location. Note that the "not logged in" location is actually put into effect when you logout of the user interface. So the default status of the not logged in location is set when you logout.

After you have all the settings for the location profile the way you want them, click on the OK button and it will save the settings for this location. Note that this information is saved even if you click Cancel on the main user settings screen.

To access the user's voice mail options click on the "Voice Mail Handling" button. The voice mail settings screen will be displayed:

The voice mail delivery and notification options are:

Voice mail to mailbox

Check this box if the user wants their 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 email

Check this box if the user wants their 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 their 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 email

Check this box if the user wants a notification email sent when they receive a new voice mail. The notification email includes a subject line with the caller ID 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 the notifications 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 by phone

Check this box if the user wants the system to call them 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. 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 the user wants the system to send them an SMS text message to notify them of a new voice mail. In the box next to this checkbox enter their SMS address, usually in the form phonenumber@provider. The text message will contain the same information that an email notification contains. Multiple SMS addresses can be specified and must be separated by a semi-colon or comma. The total length of this entry cannot exceed 1024 characters.

From address for emails

If you have chosen any option that causes the system to send an email, 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 the user wants missed calls to create a voice mail with the caller's information. The audio of the voice mail will only say "missed call". 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 setting determines what script will be run when the user press the * on their phone. The options are:

- user_menu, which is the most common choice.

- user_menu_secure, which forces the user to enter their PIN each time they enter the user menu.

- user_menu_no_group, which does not include the group options that are in user_menu.

- user_menu_secure_no_group, which forces the user to enter their PIN and does not have the group options.

On-hold script

> This specifies the script to run on calls that this user has on hold. If it is set to <none> the the on-hold script for the group that the call belongs to will be used.

Dialtone replacement script

> This setting only works when the user is associated with a MGCP phone. When the phone is taken off hook, instead of presenting a dial tone, the script specified here is run. You usually want this set to <none> so that the phone works normally.

The options for callers leaving the user a voice mail are:

Idle rings

> When the user is available for a call, this is the number of rings without answering before the call will be sent to their voice mail.

All VM callers sent to generic script

> Select this option if the user wants to have all their voice mail taken the same way regardless of their status.

Generic script

> Choose "vm_generic" unless a script has been customized for you. This script will play the prompt "vmgeneric". To record this prompt, the user must press * on their phone and in the user menu select 3 for recording voice mail greetings and then 1 for the generic greeting.

> The vm_generic script will first play the user's greeting. The caller can press 0 to go back to the operators, so the user can say that in their greeting if they want. Next a prompt with instructions for the caller leaving voice mail is played. A beep is then played and the caller records the voice mail. When they are finished they can either hang up to send the voice mail or press the # key for more options. If the caller presses the # key at the end of recording their voice mail, they will be given a

menu that allows them to listen to the message by pressing 1, re-record the message by pressing 2 and deliver the message by pressing 3 or hanging up. If they press 4, it will cancel the voice mail.

VM callers sent to script depending on user's status

Select this option if the user wants to have their voice mail handled differently depending on their status. These scripts mainly differ in the greeting they play, allowing the user to indicate their status in the greeting to the caller leaving a voice mail. If the user is not going to use the FacetPhone user interface, they should not use selective voice mail handling.

No answer script

If the user is using selective voice mail handling, this setting selects the script to run when they are available for calls and their phone is idle. If the call exceeds the number of idle rings set above with no answer, then it will go to their voice mail using the script specified here. Choose vm_idle unless a script has been customized for you. The standard vm_idle script will first play the user's vmidle prompt. To record this greeting, the user must press * on their phone to enter the user menu, then choose 3 to record voice mail greetings and 2 to record the idle greeting. In all of the voice mail handlers, the caller can press 0 to go back to the operators, so the user can say that in their greeting if they want. Next a prompt with instructions for the caller leaving voice mail is played. A beep is then played and the caller records the voice mail. When they are finished they can either hang up to send the voice mail or press the # key for more options. If the caller presses the # key at the end of recording their voice mail, they will be given a menu that allows them to listen to the message by pressing 1, re-record the message by pressing 2 and deliver the message by pressing 3 or hanging up. If they press 4, it will cancel the voice mail.

Busy script

If the user is using selective voice mail handling, this setting selects the script to run when they are on the phone and do not have call waiting turned on when a new call rings to them. Choose vm_busy unless a script has been customized for you.

The standard vm_busy script will first play the user's vmbusy prompt. To record this greeting, the user must press * on their phone to enter the user menu, then choose 3 to record voice mail prompts and then 3 to record their busy greeting. Since this is their busy greeting, they might want to say in this greeting that they are on another call. After their busy greeting, a prompt with instructions is played, then a beep and the voice mail is recorded. After recording the voice mail, callers will have the same options as with the idle script.

Out script

If the user is using selective voice mail handling, this setting selects the script to run when their status indicates that they are not at work. Choose vm_out unless a script has been customized for you. The standard vm_out script plays the user's vmout prompt. To record this greeting, the user must press * on their phone to enter the user menu, then choose 3 to record voice mail prompts and then 4 to record their out greeting. They may want to include in this greeting that they are not currently at work.

Unavailable script

If the user is using selective voice mail handling, this setting selects the script to run when their status indicates that they are at work but not available for calls. Choose vm_unavail unless a script has been customized for you. The standard vm_unavail script plays the user's vmunavail prompt. To record this greeting, the user must press * on their phone to enter the user menu, then choose 3 to record voice prompts and then 5 to record the unavailable greeting.

Defining Scheduled Events for the Auto-attendant Operation

You can define scheduled events to turn the auto-attendant on or off or change which auto-attendant is in effect.
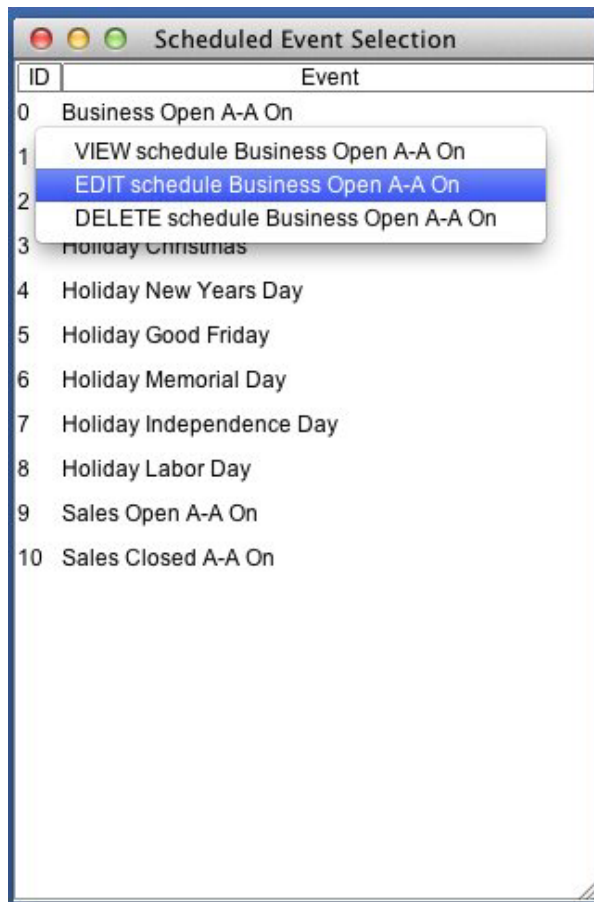
To define a new scheduled event, on the FacetPhone Administrator menu choose:

Add -> Add a Scheduled Event

If you are modifying an existing event choose:

Configure -> Configure Scheduled Events

If you are configuring an existing event, you will choose it from a list:

Whether you are adding a new event or have selected an existing event to edit, then the Gateway Configuration screen will be presented:

The settings on this screen are:

Event name

> This is a descriptive name for the event that makes it easy to identify from the list of events.

"Event time of day" settings spedify the time that the event will occur and whether it will override another event that has the same time.

Hour in 24 hour format

Enter the hour when the event will occur in 24 hour format. For example if the event occurs at 5:00PM, enter 17 here.

Minute

Enter the minute of the hour when the event will occur. For example, if the event occurs at 5:00PM, enter 0 here.

Override other events occurring 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

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

Monday – Sunday

Check the boxes for each day of the week that this event should occur.

Select by days of the year

Holiday override events should be set as "Select by days of the year".

Days of month

For an event selected by days of the year, enter the days of the month that the event will occur separated by commas. For example, if the event is meant to override the normal open event for Dec 24th and 25th, then enter 24,25 here.

Months of year

> For an event selected by days of the year, enter the month(s) that the days enter above will apply. For example, if the event is meant to override the normal open event on Dec 24th and 25th, then enter 12 here. You can enter multiple months here although it is rarely needed. However, if you wanted an override event at the time of the company open for the 1st day of each month, you could enter 1 in the days field and 1,2,3,4,5,6,7,8,9,10,11,12 in this field.

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.

Auto-attendant Setup

FacetCorp support engineers will help you create your auto-attendant(s) that work according to your specifications. The part that you will most likely need to contribute is the recording of your prompts. The easiest way to record a prompt for the auto-attendant is with a phone on the FacetPhone system. A user who is an operator or an administrator for the system will have an "Operator" item on the main menu of the FacetPhone user interface. FacetCorp support will have configured the names of the prompts you need to record in the ivr.local.cfg file. To record one of them, on the menu choose

Operator -> Record system-wide voice prompts

There is another item for recording group voice prompts. This is only used when prompts of the same name need to be recorded for more than one group, such as the group_vmgeneric prompt which is the group's voice mail greeting. After you choose to record a system-wide prompt, a screen will be presented to choose the prompt to record:

Choose the prompt you want to record and then click the Record button. The phone associated with the user interface you are using will ring or automatically go off-hook. It is best to pick up the receiver to reduce background noise in the recording. Follow the instructions on the phone. Note that you must explicitly save the prompt. If you just hang-up when you are through speaking the prompt, it will not be saved.

When you choose to record a group prompt, the procedure is the same except that you choose the group in addition to the prompt name.

You may also have prompts professionally recorded. In order to be used on the FacetPhone system, the format of the recording must be a wav file sampled at 8000 Hz, mu-law, mono, 8 bits. We have also found that most prompts that have the right volume to be played on computer speakers need their volume reduced to 30% of the original to be the correct volume for the phone.

Setting Up a Conference Bridge

To setup a conference bridge on a FacetPhone system, you create a new user, make that user never available for phone calls and set the user's voice mail handler to "conf_bridge_on". This allows users to simply call the conference bridge user's extension to join the conference. A DID can be setup to point to this user if you want outside callers to be able to directly enter the conference bridge. Otherwise, a user can transfer outside callers to the conference bridge extension to put them in the conference.

Begin by adding the new user:

All you need to specify on the main user configuration page is the extension number, user name and first and last names. If you want the conference bridge to be PIN protected, enter a PIN for this user. Anyone entering the conference bridge will be prompted to enter the PIN. Next check the "not logged in" location to be sure the user has no phone defined:

Nobody will login as this user, so if the "not logged in" location has a Voice type of None, then all calls to this user's extension will go to voice mail. The final step is to set the voice mail handler for the conference bridge user:

All that needs to be set on this screen is to select "All VM callers sent to generic script" and the choose "conf_bridge_on" for the Generic script. The conf_bridge_on script allows the caller to enter the conference associated with this extension.

Using the facetphone Command

FacetPhone is started and stopped using the command prompt on the FacetPhone server to run the facetphone command. This section reviews the usage of the facetphone command. If the server is running multiple instances of FacetPhone, then for each of the commands below, you would add at the end the name of the instance as it appears in the /etc/facetphonedirlist file.

facetphone start

This command starts the FacetPhone servers in the standard way. The server will control the branches that it has already been configured to control.

facetphone stop

This command stops the Facetphone system. All the phones and lines are disable and everyone is logged out of the FacetPhone user interface.

facetphone status

This command will report whether the FacetPhone servers are currently running or not.

facetphone restart

This command will cause the FacetPhone servers to restart. You might want to do this if you have change a setting in sys.local.cfg that requires a restart to take effect.

facetphone reload

This command will cause the FacetPhone system to re-read the configuration files. If you have made a change to one of the configuration files that must be changed manually and the setting change does not require a restart, then you use this command to cause the settings to take effect.

facetphone local

This command will start the FacetPhone servers in a mode where the system only controls the local branch where the server is. You would use this on a backup server at a branch that has lost its connectivity to the main FacetPhone server. This would allow the local server to control just its local branch.

facetphone master

This command is used on a backup server to have it take over the operation of all branches.

facetphone giveback

This command is used on a backup server when it is serving as the active server and you want to give back control to the master server. This will cause the backup server to stop running FacetPhone and go back to syncing with the master server.

facetphone branch

This command is used to give you the ability to select each branch that the server will control. After you select which branches to control, this command will reload the FacetPhone configuration if the servers are running and start them otherwise.