FacetPhone V5 Release Notes

FacetPhone V5 -- Changes Since V4
December 2008

- Added a built-in IVR function:

  int Prompt_exists(char *prompt)

  This function takes the base prompt name with no leading path and no file name extension and searches the normal prompt search path to see if it exists in any of the seach path locations (user, group, localsystem, system vmprompts directories).

- Added a built-in IVR function:

  void Get_base_prompt_name(char *path, char *basename)

  This function takes the full path name of a prompt including its file name extension and returns the base prompt name with no leading path and no file name extension.

- Made a change to allow the display_did_party_as_cid setting in sys.local.cfg to also be specified individually for each line. When the value of this setting is 'y', the called group name will replace the caller ID name in the caller ID displayed on the user interface and on the telephone's caller ID display. The sys.local.cfg value is used as the default value for each line. When setting in the lines.cfg file, the setting must be manually edited into the file and the servers restarted.

- Made a change to allow the put_called_group_in_vm_subject setting in sys.local.cfg to also be specified individually for each line. When the value of this setting is 'y', the called group name will be put as the subject for voice mails left by callers who came in on the line. The sys.local.cfg value is used as the default value for each line. When setting in the lines.cfg file, the setting must be manually edited into the file and the servers restarted.

- Added support for the Transfer key on the Polycom IP phones using the soft keys to indicate the type of transfer.

- Added a feature to allow any admin user's PIN to be used to login as any user. In order to turn this on, add the line to sys.local.cfg:

  <allow_admin_pin_master=y>

  FacetPhone must be restarted for this setting to take effect.

- Added a command that can be invoked by fp_client using the "raw" command to start an IVR that can make an outgoing call:

  !<start_ivr_call><script=x><user=y><group=z><dial_number=99729859901>

  The user specification can be left out and the script will run for the user logged in with fp_client. The group specification is the group that the script should run as with the default group specified as _default_ . The dial_number is the number that will be available in the script in the DATA1 built-in variable. It should include the leading 9 or 8 plus trunk group.

- Added a setting to allow a single key unpark on Polycom IP phones. The setting is in sys.local.cfg and is:

  <allow_single_key_unpark=y>

  The default is for this setting to be turned on. When it is turned on and there is only one call parked, the unpark button will immediately pick it up. If there is more than one parked call, then it will prompt the user to enter the call ID. When the setting is turned off it always prompts the user to enter the call ID.

- Made a change to the feature controlled by the <display_did_party_as_cid=y> setting to change the called group that is displayed on the caller ID when the group is changed by an IVR process.

- Added a setting to specify a time when all phones should be rung in order to send a caller ID string to them that will set the time on the phone. Polycom IP phones are excluded since they do not set their time from a caller ID time. The phones are rung one at a time for 10 seconds each. The default time is 3AM. This may be changed with the setting in sys.local.cfg:

  <set_time_on_phones_hour=3>

  To turn the feature off, set the value of this parameter to 24 or greater.

- Changed the font requested by the user interface from Helvetica to Arial for more reliable selection of a good font on Windows.

- Added support for distinctive ringing on Polycom 501 IP phones.

- Added a new on-hold script music_on_hold_ring_back that is hard coded to ring back the operators of the call's group after one minute of being parked. It does not ring back for calls on hold.

- Added a new voice mail handling IVR script for use as the idle script when selective voice mail handling is being used. The script is vm_idle_park_page and it gives the user the choice of pressing 1 to leave voice mail or 2 to have the call be parked and have the called party be paged. The user should record their idle greeting to say something like:

  Hello, this is _______
  I am in the office, but missed your call.
  To leave me a message, press 1 and record it after the beep.
  To have me paged, press 2.

  The page will say "Call <callid> is parked for <called user's name>"

- Added a new built-in variable available to IVR scripts CALLID which is a string representation of the call ID of the call the script is running for, such as "1".

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- Added a feature to have all analog phones rung in the middle of the night in order to get the time set properly on them. The default is to ring all the phones at 4AM. To change this, enter the setting in sys.local.cfg:

  <set_time_on_phones_hour=n>

  If n is greater than or equal to 24, then the feature is turned off.

- Added a feature to reset all Voipack and Adtran gateways in the middle of the night. By default, this is done at 3AM. To change this, enter the setting in sys.local.cfg:

  <reset_hour=n>

  If n is greater than or equal to 24, then the feature is turned off.

- Added a new gateway manufacturer type AC_MEDIANT_V2 which is listed in the user interface as "Audiocodes Mediant T1 V2". This manufacturer type differs from the original Audiocodes Mediant T1 type (AC_MEDIANT) in that the name on the original version has square brackets around its IP address for the name, and the V2 version has only the IP address for the name without the square brackets.

- Added the called group name to the information displayed on a Polycom IP phone when an outside call is ringing to the phone.

- Supported distinctive ringing on Polycom IP phones.

- Added a sys.branch.cfg configuration file that is used to hold only those items that are unique to a branch. This allows the other settings in sys.local.cfg to be copied to a backup server without disturbing the settings which need to remain unique for the backup server.

- Made a change to allow distinctive rings to be set to a value of -1 which will cause a generic ring request to be sent to the phone instead of a particular ring type. On a Polycom phone this allows the ring type that has been selected on the local phone menu to be used.

- Made the maximum parties per call into a configurable parameter. The default is 50. For sites having many Polycom phones that will be paged to, this parameter needs to be increased to the maximum number of phones that will be paged to. For example, to set this parameter to 100 parties per call, add to sys.local.cfg:
  <max_parties_per_call=100>

- Increased the size of the voice mail recipient to accept a list of email addresses.

- Added a system-wide setting which determines if calls can be transferred to a phone that is offhook not connected to a call. The setting is off by default. To turn it on, add to sys.local.cfg:
  <allow_xfer_to_phone_offhook=y>

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- Changed the IVR script language processor to accept a define maximum number of subroutine arguments instead of being hard coded for 4. The maximum number is currently defined as 20.

- Removed the restriction of a maximum number of voice mails in a folder.

- Added new IVR functions for playing a prompt that will not be interrupted by any digits dialed by the caller:

  Play_prompt_no_stop(char *prompt)
  Play_prompt_no_silence_no_stop(char *prompt)

- Added a setting that determines whether the inside parties to a call will be considered in determining the groups that the call is associated with when displaying in group call windows on the user interface. The setting should be put in sys.local.cfg and is:

  <use_inside_party_for_call_groups=y>

  The default value is 'y' to consider the inside parties in the group associations.

- Changed the default for the acd_oper_change_stat_no_answer feature to be:

  <acd_oper_change_stat_no_answer><old=oper on duty><new=at desk>

  Previously it had been turned off by default. To turn it off add to the sys.local.cfg file:

  <acd_oper_change_stat_no_answer><old=><new=>

  This setting needs to be on when Automatic Call Distribution is being used in order for the system to work properly when an operator is not able to answer calls and has their status set to receive ACD calls.

- Changed the PBX to not have silence suppression turned on when an IVR is communicating with a phone. It is still turned on when an IVR is communicating with a line.

- Added an auto-attendant model script that can be copied and parameters set to easily implement customer auto-attendants. The model ivr script is in a-a-545.ivr and documentation for each of the parameters is in a-a-545.doc. These are installed in:

  /usr/facetphone/ivr/system/ivrscripts

- Made a change to treat an RSIP with a "forced" reason the same as a "reset". The EyepMedia phone sometimes sends this.

- Made a change such that a FacetPhone system will only communicate with the gateways at a branch for which the system is configured to be in control of the gateway's branch.

- Added a feature in which an outgoing caller ID number and name may be specified per user for calls going out on a T1. The name may not work with all T1 services.

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- Added a feature where an outgoing caller ID name may be specified for a trunk group. This is overridden if a user name is present. The name may not work with all T1 services.

- Added a system-wide setting that determines whether the "record all calls" feature applies to outgoing calls. The setting is in sys.local.cfg:
  <allow_record_all_outgoing=y>
  The default value is to allow recording of outgoing calls. Change the value to "n" to disallow automatic recording of outgoing calls.

- Added a feature that allows control of playback of voicemail on a phone. While playing a voice mail, pressing 0 will pause/resume the playing of the voice mail. Pressing * will backup play by 4 seconds and pressing # will advance play by 4 seconds.

- The user interface was modified to run properly on Java 1.6 versions.

- Added support for the SIP OPTIONS request. This is used to support backup proxy server operation of the Audiocodes Mediant 1000 T1 gateway.

- Added support for ITSP service.

- Added three new commands to the facetphone script:

  facetphone takeover local | master
  facetphone giveback
  facetphone branchctrl

  These new commands are used under the following circumstances:

  A backup server is taking over operation for only its own branch. The "facetphone takeover local" command is used to do this. The master computer should not currently have connectivity or control of the local branch when this is done.

  A backup server is taking over operation for multiple branches. The "facetphone takeover master" command is used to do this. For each branch defined on the system, the user will be prompted to enter y or n regarding control of the branch. Each question must be answered before FacetPhone is started.

  A backup server is being shutdown prior to a master server regaining control of the branch. The "facetphone giveback" command on the backup server is used to do this. Voice mail that is new on the backup server will NOT be automatically merged back into your main server. Please contact FacetCorp support for help with this task.

  A master sever is changing the branches that it is controlling (either taking over branches or giving up control of branches).

The "facetphone branchctrl" command is used to do this. If FacetPhone is already running, it will be signaled to re-read the new configuration. Otherwise it will be started with the new configuration.

- Added the fp_branchctl program used by the facetphone script to create the proper branch.cfg file based on which branches are chosen to be controlled by the server.
- Made changes that allow a master system's branches that it controls to be changed without restarting FacetPhone.
- Added "Fixed order" ACD processing where the same operator is always chosen first if available, working down a fixed order to ring the other operators if the first is busy.
- Made the "Session" number in the log of instant messages (IM_log.txt) be a unique serial number in the same pool as unique call serial numbers in the call detail records. This allows easier identification of IM sessions in the log.
- Made changes to keep the server from answering devices at a branch that the server is no longer controlling.
- Made a change to have the stutter dialtone not indicate new voice mail. It previously meant that you had new voice mail and/or had one or more calls on hold. Now it will only mean that one or more calls are on hold.
- Made a change to blank pad caller ID strings sent to an Adtran gateway.
- Made a change to display the number of calls on hold on the screen of a Polycom phone.
- Changed the SIP trunks to use the Call-ID rather than the IP address to identify dialogs.
- Changed the SIP trunks to register with expires 0 to cancel the registration if all of its lines have been disabled.
- Added the ability to specify the Polycom button map per phone. As of this build it must be hand-edited into the phones.cfg file with an element such as:

  <special_button_map=PC501_LN_AGENT>

- Added a new Polycom button function to set a user's status to a particular value. In the button map, these are indicated as Status 1 through Status 8. In the sys.local.cfg file these are related to valid status definitions on the system such as:

  <bp_status_button=1><ustat=Training>
  <bp_status_button=2><ustat=Team Lead>
  <bp_status_button=3><ustat=Outbound>
  <bp_status_button=4><ustat=Personal>
  <bp_status_button=5><ustat=Break>
  <bp_status_button=6><ustat=Lunch>
  <bp_status_button=7><ustat=Wrap>

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<bp_status_button=8><ustat=Ready>

- Added a new IVR function:

  Transfer_to_group("groupname");

  This will cause the IVR process to exit and the call will be put through the normal incoming call processing for the group. If the group is currently setup to run an auto-attendant, then it will be run. Otherwise the operators for the group will be rung.

- Made a change to prevent the RECORDING message from being displayed on Polycom phones if the sys.local.cfg setting hide_recording_on_ui was set to "y".

- Added a system-wide setting that specifies that lines are selected for outgoing calls in either ascending or descending order. This is specified in sys.local.cfg as:

  <choose_outgoing_line_decending=y>

  If the value is 'y', the highest numbered available line will be selected. If the value is 'n', the lowest numbered available line will be selected.

- Added the new fp_aagen application that builds an auto-attendant IVR script from a configuration file and a master auto-attendant model script. See /usr/facetphone/ivr/system/a-a.doc for information on using this new application.

- Made a change to prevent a new call being sent to a Polycom IP phone while the user is in the middle of a transfer using the Transfer button on the phone and associated soft keys.

- Made a change to prevent the log file from exceeding 2GB.

- Changed the acd_oper_change_stat_no_answer_old and acd_oper_change_stat_no_answer_new settings to have no default values.

- Made a change to not report each voice mail deletion to the PBX process during a bulk voice mail purge. Instead the new count is reported after all the voice mails in a folder have been purged.

- Made a change to put the called group in the voice mail subject of a call recording if the put_called_group_in_vm_subject is turned on. Previously this setting only applied to voice mails and not call recordings.

- Made a change so that a manager in a group can pickup any call that an operator for the group can.

- Made a change to simply reply to gateways with a known name but wrong IP address instead of matching them by name.

- Introduced a new version of the call detail record:
  New version number "CDR_2.0".

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Serial number of the call is unique.
The DID field is now filled in.
Groups that the call progresses through are logged as call parties.
Parties are in the record in the order they entered the call.

- The fp_db program now starts the fp_mysqldb program if it exists. The fp_mysqldb program will then make a TCP connection to the fp_db program and read call detail records from it. The call detail records will be stored in a MySQL database in a future release.

- Added a user status table to the FacetPhone MySQL database. The user status changes that are written to the user_status.txt file are now also added to this table.

- Made a change to the fp_mysqldb program to keep its connection to the MySQL database daemon alive during periods of inactivity.

- Added a sys.local.cfg setting that determines whether the MySQL database will be used to hold call detail records:

  <write_cdr_to_mysql=y>

  The default is NOT to write to the MySQL database. Therefore anyone wanting this feature must have it turned on with this setting.

- Made a change to have the Polycom IP phones use RFC2833 to encode DTMF if both it and the other endpoint's gateway are configured to use RFC2833 (use_rfc2833=y in gateway configuration).

- Changed the config_report.txt to the have the correct values for the gateway settings use_audio_digit_collection and use_rfc2833. Also changed it to not print the itsp_password.

- Made a change to indicate the voice mail recipient in the call detail record.

- Made a change to automatically hangup a Polycom IP phone when it is left offhook at the end of a call.

- Added support for the Cisco 7940 and 7960 IP phones with MGCP firmware. All of the functionality that is supported on the Polycom IP phones is supported on the Cisco phones. In addition a conference display is supported in which up to 6 of the conference parties are shown on the Cisco display.

- Made a change to disallow starting recording by a party that is currently on hold by the other party in an extension to extension call.

- Increase allowable number of SIP headers.

- Made a change to adjust a user's default status for the not logged in profile to not have the IM availability turned on.

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- Reduced number of commands being sent to a Polycom phone when setting the transfer display.
- Added a field to the call detail record that has a value of "y" or "n" and indicates whether the call was ever queued. The version of the call detail record was changed to 2.1.
- Made a change to drop an unresponsive endpoint from a conference call or page to IP phones instead of taking down the entire call.
- Made a change to not use the Type Of Service mark on packets being sent to a Cisco IP phone. It appears to ignore voice packets that have the TOS byte set.
- Made a change to allow extensions to be greater than 4 digits.
- Changed the default for auto-offhook on intercom calls to be off.
- Made some enhancements to the auto-attendant generator. See /usr/facetphone/ivr/system/ivrscripts/a-a.doc for information about the settings for the current version of fp_aagen.
- Changed the standard voice mail IVR scripts to return to the operator group if the caller stays on the line after leaving a voice mail. If an auto-attendant is turned on for the group, then the caller will be put into the auto-attendant. If not, then the call will ring to the operator group.
- Changed the standard company directory IVR script to offer the option to return to the previous menu.
- Added SIP trunking settings to the gateway edit screen.
- Added "Ring for transferred calls" group setting to specify a distinctive ring for calls that have already been initally answered but are ringing again because the call is being transferred, ringing back while parked, etc.
- Added "Parked call ring back seconds" group setting to specify the number of seconds a call is parked before it rings back to the operator group. This setting is used in the standard music_on_hold script now. It should be noted that the time is only checked between songs, so the ring back time will not be exactly what is specified but will occur the first time a song has ended and the ring back time has been met.
- Added gateway settings to specify how digits are detected during a call. "Digits detected in audio" should be checked if the gateway will not detect the digits and the audio should be examined to detect the digits in the audio stream. "RFC2833 digit detection" should be checked if RFC 2833 RTP payload packets for DTMF digits will be used. If neither of these are checked, then it is assumed that the gateway can detect the digits and will notify of them.
- Added user specific outgoing caller ID number and name settings. These will only be used when the outgoing call is made on a T1. These settings are viewable by the user but can only be set by the administrator.

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- Added Cisco 7940 and 7960 as a type of IP phone that can be selected.
- Added Grandstream GXW410x as a type of gateway that can be selected. This is a SIP FXO gateway.
- Added a user location profile setting that allows the dialtone to be replaced with the running of an IVR script for special purposes.
- Made a change to the #0 handling to only pickup calls ringing at the local branch.
- Added a method for a manager to temporarily turn recording on and off for the current call of an agent. The rules for the manager being able to do this are the same as for monitoring and barging in. The manager can click on the call in a call display and one of the options will be to record or stop recording the call. Note that the recording will be for the agent and will therefore be put in the folder where the agent's recordings go.
- Added missed call "voice mails" when a caller hangs up on your voice mail before leaving a message. The audio of the missed call entry says "missed call". The caller ID and date and time of the call will be stored as with a normal voice mail. Also the normal voice mail handling will be used for the missed call entry. The current implementation does not create a missed call entry if the caller hangsup before entering your voice mail. The standard voice mail scripts have been changed to implement this feature. Any custom voice mail handling scripts must have calls to the new Add_missed_call() function in order to implement this feature.
- Made a change to allow a call in on-hold music or message be properly transferred to an operator.
- Made changes to the user interface to have the group names sorted properly in drop lists.
- Added the script reboot_pc501 to the FacetPhone bin directory. Usage: reboot_pc501 [all | STATION [STATION2 [...]]]
- Finished the missed call feature. A missed call entry will now be created if the caller hangs up before entering the called party's voice mail.
- Made a change to the Polycom single key unpark feature such that a sole parked call on the system will not be picked up if its parked party is on a phone or line from another branch.
- Made the default keep-alive message interval be 5 minutes. Previously the keep-alives were disabled by default.
- Made the inclusion of missed calls in a user's voice mail an option. This is set with the user interface on the user's Voice Mail Handling screen.
- Made a change to keep re-trying an out of order phone or line forever instead of disabling it.

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- Added the conference bridge feature. In order to setup a conference bridge you create a dummy user who is never available for a phone call and whose voice mail script is set to conf_bridge_on. Callers who want to join the conference dial that user's extension. A DID for a conference bridge can be used to direct calls to that user. If you want a conference bridge turned off, the dummy user's voice mail script can be set to conf_bridge_off. You can also temporarily make your own extension be a conference bridge by setting yourself unavailable for phone calls and setting your voice mail script to conf_bridge_on. The first party to a conference will hear on-hold music until the second party joins. As each party joins, a tone will be played to let the conference members know that a new party has joined.

- Changed the handling of a SIP line to send a CANCEL instead of a BYE if the call is being terminated before the INVITE has been answered.

- Made a change to prevent a null user name from being used in the From: and Contact: headers. This could cause a line to be shown left in use when it wasn't.

- Made enhancements to the fp_aagen program for generating auto-attendant IVR scripts:

  For systems where multiple FacetPhone instances are installed, the instance name as specified in /etc/facetphonedirlist can be specified as an additional argument to fp_aagen.

  The model name may be specified in an auto-attendant .cfg file to use a model other than the default "a-a.model". The model name specified will have ".model" appended to it to create the name of the model file.

  Made the bilingual support more complete.

  Added a a-a-sub.model to be used as a model for an auto-attendant script that is called from the top level auto-attendant with the alternate language setting as an argument.

- Changed entry of MAC addresses to ensure that they are 12 characters long.

- Added ability for user interface to provide SIP trunking settings.

- Added the ability for the IVR process to detect silence instead of having silence suppression turned on for the gateway port.

- Changed the default for including missed calls in voice mail to be off.

- Added the IVR variable CALLER_UID that contains the user extension of the caller if they have authenticated themselves to the script.

- Made a change to the logic to dequeue calls when an operator becomes available such that it will dequeue calls from the groups round-robin if the operator is an operator for multiple groups that have queued calls.

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- Adjusted the dial tone detection while recording voice mail from 60 ms to 1000 ms.

- Changed the non ITSP caller id creation to work with the "use_lines_from_trunk_group=" directive. This is used to make a trunk group give an outgoing caller ID without requiring that the trunk group has an individual set of lines.

- Added the built-in variable LANG to the IVR scripting language. If LANG is 0 then the default language prompts should be played. If LANG is a positive number then it can be used to concat the names of prompts that use the LANG number. The LANG variable will be 0 when the call is started and is only set under the direction of an IVR script (see next item). The LANG variable stays with the call so that the IVR process that sets it can end, as in transfer_to_group, and the next IVR process running on the call will begin with the same LANG setting that the previous IVR process left it with.

- Added the built-in function
  Set_lang(n);
  which will set the LANG variable for the call to the integer value provided as an argument.

- Added a sys.local.cfg setting:

  <limit_to_oper_group_call_disp=y/n>

  If this is set to "y", then calls will only appear in the call display for the group that is currently the operator group for the call. When this setting is "n", then the group memberships of the parties in the call will also determine which groups' call displays the call appears in. The default value is “n".

- Added a sys.local.cfg setting:

  <pref_cid_number_over_name=y/n>

  If this is set to "y", then the CID number will be displayed on the screen of a Polycom phone instead of the CID name after the call is answered. The default valune is "n".

- Made the "phone login" location profile editable so that users can specify their default status to be used when they login to a phone with the user menu. When editing this profile, leave the voice type set to None. When you login in at a phone, the voice type will automatically be changed to "Station" and the station number set to the one where you logged in.

- Made a change to the administration of IP phones to prevent leaving the IP address blank when the dynamic address box is not checked.

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