PLATON

IP PBX

Administrator's Manual

For CS-2200 / CS-3200 / CS-4200

Version   : 1.2
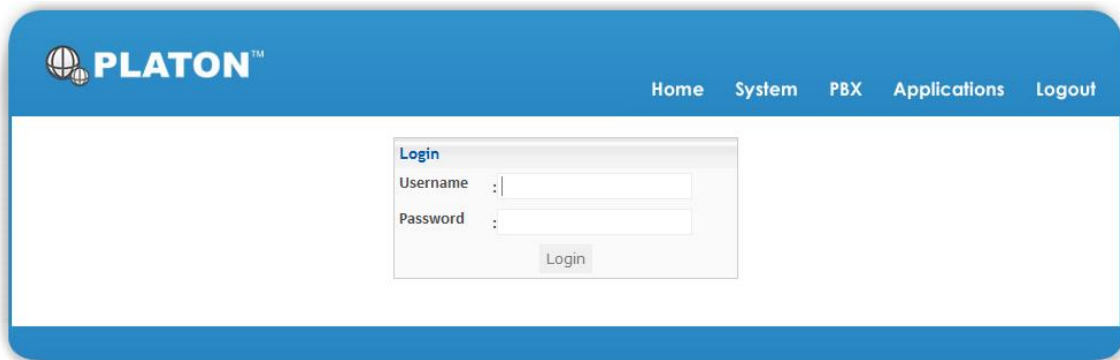
Date   : 15 Jun 2017

Table of Contents

Revision History

Date	Version	Description
2 Mar 2016	1.0	Initial Version
17 Mar 2017	1.1	Corrected typo errors in some pages
15 Jun 2017	1.2	Added Revision History

Login Screen

1) Type in Username and Password (Case sensitive)

2) Press "Login" button to Login\

** System will lock the user account if user input a wrong password for more than 3 times, please use "User Management" module to unlock the user.

 Backup and Restore

System create backup for sytem configurations and database data, users may also create backup or restore data through this module.

Create Backup / Restore from backup

1) Backup: Click on "Backup Now" located at top part of backup list table

   Restore: Click on "Restore" on the desired restore date.

2) Select the items need to backup or restore

- Config Files   : System Configurations
- Voicemail Files   : User's Voicemail
- Recordings   : Voice Logs (including Queue Voice Logs if available)
- Database   : System Database (e.g. User List, Voicemail Accounts…etc.)

License

License module shows the license installed on the system

Users License   : License for connecting IP Phone (including desk phone, soft phone, mobile apps..etc.)
Each device requires one individual extension, each extension requires one individual
license.

ACD Module   : License for ACD (Automatic Call Distribution) module, features including:
- ACD Agent Status
- ACD Wallboard
- Agent Activity Log
- Agent Attendance Report
- Agent Login/Logout
- Agent Management
- Agent Statistics
- Queue Management
- Queue Statistics
- Queue Voice Logs

CRM Integration module: License for CRM Integration (Call popup with contact matching fetures)

System Information

System Information							
Uptime	14:57:23 up 667 days, 14:57, 0 users, load average: 0.10, 0.04, 0.01						
Memory		total	used	free	shared	buffers	cached
	Mem:	2066872	1896124	170748	0	177832	1048872
	-/+ buffers/cache:		669420	1397452			
	Swap:	2066872	1896124	170748			

Hard disk	Filesystem	1M-blocks	Used	Available	Use%	Mounted on
	/dev/md2	145367	12611	125253	10%	/
	/dev/md0	494	27	442	6%	/boot
	tmpfs	1010	0	1010	0%	/dev/shm
	/dev/sdc1	1876	1876	0	100%	/flash

MySQL Status	Uptime: 57682590
	Threads: 4
	Questions: 48651818
	Slow queries: 1
	Opens: 4024
	Flush tables: 1
	Open tables: 64
	Queries per second avg: 0.843

System information module displays the most current system status and information such as Uptime, Memory usage and Internal Storage usage.

User Management

User Management module manages users for logging into the Platon® Web Interface.

Create/Add User

1)   Press the "Add User" link located at top part of the user list table

2)   Input User's information including Name, Username, Password and select user's permission
     (Hold "Ctrl" key to select multiple items in Permissions section)

3)   Click "Submit"

Edit User

1)   Press "Edit" link for the appropriate user and modify the user's information or permission

Unlock User

1)   If user tried to login with incorrect password for more than 3 times, system will lock the user's account, to
     unlock, click on "Unlock" link for appropriate user.

 Account Codes

To protect from IDD abuse, Account Code can be setup for each user, system will prompt for account code when user tries to dial IDD.

Create Account Code

1) Press "Add Account Code" link located at top part of the user list

2) Input User's information and Account Code and click "Submit" buttoon.

- Department : The Department name for the user (This field is Optional)
- Dept. Code : Department Code for the user (This field is Optional)
- Account Name : Select the appropriate user from the list, if user is not within the list, please add a user in "VOIP Users" module
- Account Code : Account Code for the user, it should be unqiue within the whole system

Edit Account Code

1) Press "Edit" link for appropriate User

2) Change account information and click "Submit" button

** After Account Code is created, please add "Prompt for Account Code" action in Dial Plan for any IDD related dial rule

Call Detail Report (CDR)

Call Detail Report shows all the inbound and outbound call history, users may use this module to search the call history from certain critireas such as "Caller", "Destination and Date. Search results can be listed in web-based format or even exported to Excel compatible CSV (comma separated values) format.

Searching/Export Call History

1) Input the search critirea such as Caller (called from), Destination (called to) and period of call report, please note that if no search critirea, system will show all call history.

2) The table shows the call details:

Date   : Date and Time for the call made, in the format of YYYY-MM-DD HH:MM:SS

Account   : Account Name used (related to Account Code), if available

Caller   : Telephone number that initiated the call, and in blanket showing the channel used

Destination   : Called telephone number, and in blanket showing the channel used

Channel   : Voice channel used

Duration   : Total talk time for the call (including Ringing time)

Billed Duration : Total talk time for the call (excluding Ringing time)

Disposition   : Call disposition (No Answer / Answered / Busy / Failed)

3) Click on "Export to File" (located on top of the table), to export the data to Excel format file.

Call Forward Managemnet

Call Forward management

Extension	Forward Number	Forward Mode				
101		◉ Off	○ All	○ Busy	○ No Answer after	sec.
102		◉ Off	○ All	○ Busy	○ No Answer after	sec.
103	106	○ Off	○ All	◉ Busy	○ No Answer after	sec.
105		◉ Off	○ All	○ Busy	○ No Answer after	sec.
106		◉ Off	○ All	○ Busy	○ No Answer after	sec.
107		◉ Off	○ All	○ Busy	○ No Answer after	sec.
108	968862770	○ Off	◉ All	○ Busy	○ No Answer after	sec.
110		◉ Off	○ All	○ Busy	○ No Answer after	sec.
111	936107618	○ Off	○ All	○ Busy	◉ No Answer after 15	sec.
115		◉ Off	○ All	○ Busy	○ No Answer after	sec.
116		◉ Off	○ All	○ Busy	○ No Answer after	sec.
121		◉ Off	○ All	○ Busy	○ No Answer after	sec.
122		◉ Off	○ All	○ Busy	○ No Answer after	sec.
123		◉ Off	○ All	○ Busy	○ No Answer after	sec.
201		◉ Off	○ All	○ Busy	○ No Answer after	sec.
202		◉ Off	○ All	○ Busy	○ No Answer after	sec.
203		◉ Off	○ All	○ Busy	○ No Answer after	sec.
204		◉ Off	○ All	○ Busy	○ No Answer after	sec.

Submit

Administrators may control user's extension call forward setting through this page.

Extension   : Extension number to do call forward

Forward Number : Call forward destination

Forward Mode   : Call Forward Modes

- Off   : No Call Forward
- All   : Forward All Calls to specific telephone number
- Busy   : When extension is busy, call forward to specific telephone number
- No Answer : When extension rings for specific time (specified in next field) and user did not answer, call will be forwarded to specific telephone number

 Channel Status

This page shows the system active calls, users may hangup any of the listed channel by clicking on the channel link, system will prompt for hang-up confirmation, click "Yes" to confirm hang-up the specific channel

Dialplan

Dial Plan module controls all the outgoing dial rules and incoming call routes.

When performing outgoing calls, system will check the dial rules within the assigned dial plan group (extension dial plan group is assigned within the VoIP Users module), if a dial rule is matched, the system will run the commands within the dial rule setup.

When an incoming call arrives, system will check for incoming calls dial plan group (usually is "pstn" group). For analog trunk lines, the call will arrive at "s" dial rule; for digital trunk lines (including IDAP/T1/E1/SIP/IAX Trunks), digit send will be received as dial rule, e.g. if the digit send from the trunk is 630, the system will run the commands in "PSTN" dial plan group, with dial rule 630.

Dealing with Dial Plan

When the number dialed or digit sent from trunk matches the dial rule, the call will be routed within the "Auto-Run Action List" from the first command to the last command until any of the command pickup the call. In the screen example shown above, when the system received digit sent from trunk is 630, the system rings the extensions "Peter Chan (101)" and "May Wong (103)" for 15 seconds, if no one picks up, the system will ring the extension "Andy Lee (102)" for another 15 seconds, if still no one answers, the system will redirect the call to Mailbox of "105 (Alex)" with no prompt. For a complete list of Dial Plan commands and its descriptions, please refer to **Appendix A – Platon Dial Plan Command List**

Dialplan (Cont')

<u>Understanding Dial Rules</u>

Dial rules can be setup to match with several dial digits or digits send. Dial rule can be expressed as:

X matches any digit from 0-9

Z matches any digit from 1-9

N matches any digit from 2-9

[1237-9] matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)

. wildcard, matches one or more characters

Here are some examples:

Example 1 : **1XX** dial rule matches any dial digits or digits send with 3 digits and starting with the digit "1"

Example 2 : **[123XX]** dial rule matches any dial digits or digits send with 3 digits and starting with the digit "1" or "2" or "3"

Example 3 : **[13-5XXX]** matches any dial digits or digits send with 4 digits and starting with the digit "1" or "3" or "4" or "5"

Eaxmple 4: **9XX.** matches any dial digits with at least 4 digits and starting with "9"

Dialplan (Cont')

<u>Prefix and Strip Digits</u>

When dial command comes with "Prefix" and "Strip Digits", dial rule can be modified automatically to match with the command. In this example, we'll show the command "Dial Extension".

In this dial rule, [123]XX, it matches any dial digits or digits send with 3 digits starting with "1" or "2" or "3", that's mean any digits within 100 – 399. With "Dial Extension (according to dial rule)", Prefix set to "5" and no strip digits. The system will dial the extensions 5100 – 5399, since the "Prefix" setting adds the specified digit "5" in front of the dialed digits or digits send.

Here is another example, with dial rule **76XX**, prefix "1" and strip digits "2", the system will convert the dial string from 76XX to 1XX, since strip digits removes the number of digits specified (in this case is 2) from the front and add the prefix "1" in front.

Dialplan (Cont')

Day / Night Mode Switching

Day / Night Mode Switching allows the system to switch to different dial plan group to perform different actions for the calls. An example of Day / Night Mode usage is to setup different call routes for operation hours and non-operation hours, company might want to ring the receptionist's phone during operation hours and re-direct the call to IVR system for non-operation hours.

Setting up Day/Night Mode for incoming calls

1. Define two dial plan groups (e.g. pstn_day and pstn_night) for different modes
2. Go to dial plan group for incoming calls (usually "pstn" group), and click on "Day / Night Mode Switch" (located under the "Dial Plan Group" selection box)
3. A screen similar to above screen example should be shown, select the group to be switched, e.g. if you're setting up night mode, select "Switch to: pstn_night"
4. Within the "Schedule" section, select the schedule for the profile to be run. You may specify to run night mode all the time, but add on top for specified schedule to run day mode.

Example Setting:

1. Add "pstn_night" as "Always" in schedule
2. Add "pstn_day" for the schedule Mon – Fri, 09:00 – 18:30
3. Move the priority of "pstn_day" schedule on top of "pstn_night"

Features Settings

Call Park Extension : Extension number used to Park Call

Call Park Position : Define usable slot for Call Parking

Group Pickup Feature Code : Feature code for Call Pickup within the same Group

(Call Group and Pickup Group can be defined in "VoIP Users"

module)

On-Demand Recording Feature Code : Feature for starting On-Demand Recording duing conversation

Blind Transfer Feature Code : Feature Code for Blind-Transfer (for analog phones only)

Attended Transfer Feature Code : Feature Code for Attended-Transfer (for analog phones only)

IVR Menu

Menu List (Add Menu)			
Name	Description	Edit	Delete
ivrdemo	IVR Demo	Edit	Delete

To setup IVR (Interactive Voice Response) system, after a IVR menu is setup, it can be used from the Dial Plan module.

Create IVR Menu

1) Click "Add Menu" link located at the top part of the IVR menu list

2) Input the Name and Description for this IVR Menu

3) If extensions or features can be dialed within the IVR menu, please select the Dial Plan profile within "Direct Dial for" select menu.

4) Setup the Actions to be run when caller reaches this IVR menu in "Auto-Run Action List", unlimited number of actions can be added, for details for "Action", please refer to "Dial Plan / IVR Action List and Descriptions"

5) Setup the Action to be run for each key pressed, including 0,1,2,3,4,5,6,67,8,9,* and #. If caller pressed an undefined action key, it goes to "Invalid" action list, if caller didn't press any key within a specified period, it goes to "Timeout" action list

IVR Menu (Cont')

Edit IVR Menu

1) Click on "Edit" for appropriate IVR Menu within the IVR Menu list

2) Change the settings and click "Submit"

Example of IVR Menu

Trunk Management

IAX Trunk (SIP Trunk)	Add Trunk			
Trunk Name	Username	Host	Edit	Delete
Platon	abc123	voip.platonvoip.com	Edit	Delete

VOIP Trunking can be setup to connect other VOIP Telephone System, supported VOIP Trunking protocols including SIP and IAX2.

Add/Edit IAX2 Trunk

1.   Click "Add Trunk" or "Edit" in "IAX Trunk" page

2.   Input the Trunk information:

    Name   : Name for this trunk, to be appeared in Dial Plan or IVR

    Username   : Username to login to this trunk

    Password   : Password for this trunk

    Host   : IP Address or Domain Name for the VOIP System

    IAX User   : Instead of IP Address or Domain Name, connects to a IAX peer

    Register to this trunk   : Check the box to make the system always do registration to the trunk

Edit IAX Trunk

Name: Platon

Username: abc123

Password: *****

◉ Host   voip.platonvoip.com

◯ IAX User   ---------- ▾

Register to this trunk   ☐

Submit Cancel

Add/Edit SIP Trunk

1.   Click "Add Trunk" or "Edit" in "SIP Trunk" page

2.   Input the Trunk information:

Name   : Name for this trunk, to be appeared in Dial Plan or IVR

Username   : Username to login to this trunk

Password   : Password for this trunk

Host   : IP Address or Domain Name for the VOIP System

Incoming Dial Group : Select the Dial Plan group to use when there's an incoming call from this trunk

Qualify   : Show the network latency in "User Status" page within "VOIP Users" module
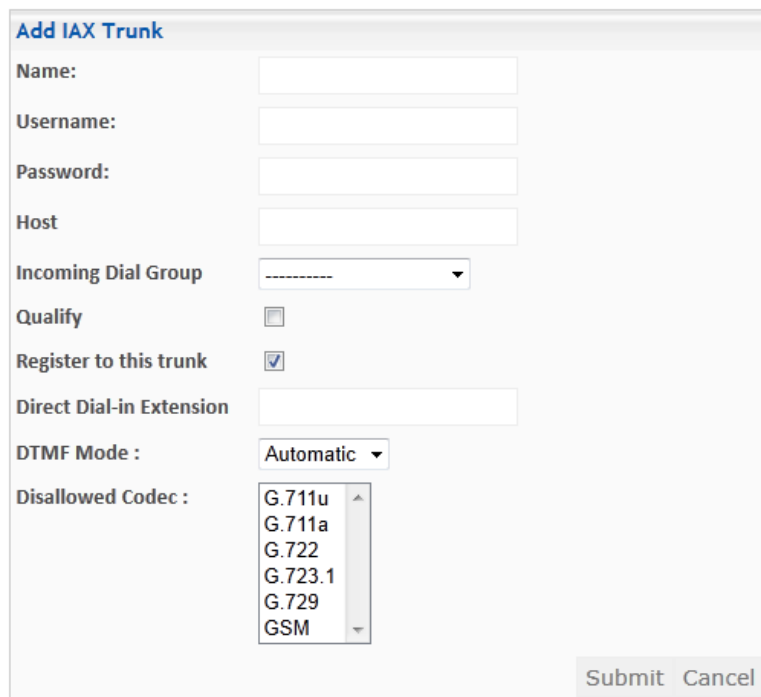
Register to this trunk   : Check the box to make the system always do registration to the trunk

Direct Dial-In Extension: Number to be automatically dialed for calls without DID number specifed

DTMF Mode   : DTMF Mode for this trunk

- Automatic : Use rfc2833 for DTMF relay by default but will switch to inband DTMF tones if the remote side does not indicate support of rfc2833

- RFC2833   : Use RFC2833 for DTMF

- Inband   :The device that you press the key on will generate the DTMF tones. If the codec is not ulaw or alaw then the DTMF tones will be distorted by the audio compression and will not be recognised.

- Info   :Use SIP method INFO or SIP INFO as DTMF

Disallowed Codec   : Audio Codecs NOT to use within this Trunk

Add IAX Trunk

Name:

Username:

Password:

Host

Incoming Dial Group   ---------- ▼

Qualify   ☐

Register to this trunk   ☑

Direct Dial-in Extension

DTMF Mode :   Automatic ▼

Disallowed Codec :   G.711u
                             G.711a
                             G.722
                             G.723.1
                             G.729
                             GSM

Submit Cancel

Voice Logs

Voice Logs module allows user to search and play back voice logged within the telephone system. Each voice files are in Windows PCM Wave (.wav) file format, which can be played by most of the audio player softwares.

Searching Voice Logs

1. Input the search critirea such as Caller (called from), Destination (called to), call duration and period of voice logs. (Note: if no search critirea, system will show all call history)

2. The table shows the call details:

    Date   : Date and Time for the call made, in the format of YYYY-MM-DD HH:MM:SS

    Caller (Channel)   : Telephone number that initiated the call, and in blanket showing the channel used

    Destination (Channel)   : Called telephone number, and in blanket showing the channel used

    Duration   : Total talk time for the call (excluding Ringing time)

    Disposition   : Call disposition (No Answer / Answered / Busy / Failed)

3. Click on "Play" to play the Voice Log directly or save into computer.

VoIP Users

VoIP Users module manages Extensions, VOIP accounts, Voicemail Boxes and Web Interface permissions for each user.

Create / Edit User (Extension)
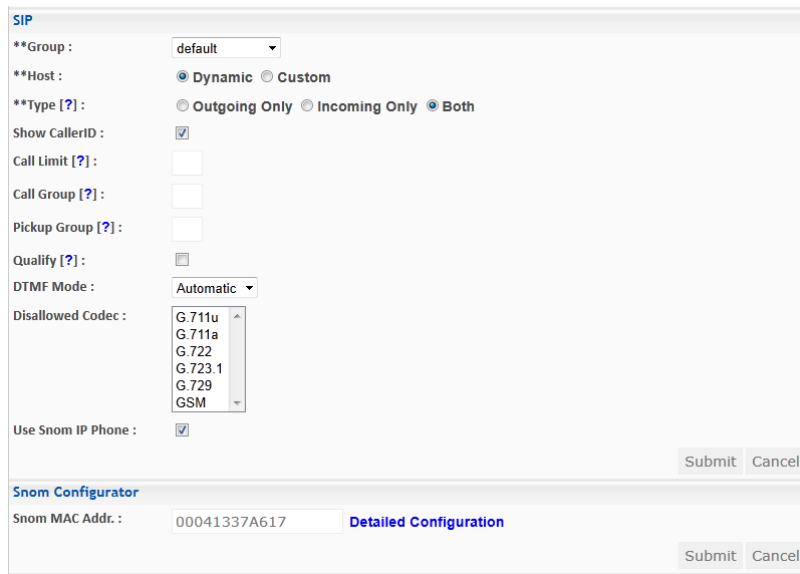
1. Click on "Add User" (located on top of user list page) or "Edit" link for the user
2. Field in User's information and check for appropriate features for the user:

   Name : Name of the User (to be shown on each Phone and Directory list)

   E-mail : Optional, Voice Mail notifications will be sent to this e-mail if entered

   User's Password : To be used in Voice Mail, Web Login, Remote Call Forward and Virtual Extension (if feature available)

   Extension/Username : Extension number for this user, also used as login user name for IP phone or VOIP software and Web Interface Login

   VoIP Password : Used as login password for IP Phone or VOIP software

   Show in Directory : User will be shown in phone directory if checked

   SIP Enabled : SIP VoIP service will be enabled if checked, SIP standard is widely used in most

22

VoIP Software

IAX2 Enabled : IAX VoIP Service will be enabled if checked, IAX standard is widely used in most IP PBX System

H.323 Enabled : H.323 VoIP Service will be enabled if checked, H.323 standard is old fashioned VOIP service used in early VoIP hardware

Voicemail Enabled : Voice Mail feature will be enabled if checked

Web Interface Enabled : Allows user to login to this web interface if checked

3.   SIP Settings:

Group : Dial plan group to be used

Host : Allow Dynamic (any IP) host or Custom (fixed IP) host

Type : Setup to allow Outgoing / Incoming / Both way calls

Show CallerID : Hide Caller-ID if not checked

Call Limit : The concurrent call (number of simultaneous calls) limitation for this user

Call Group : Defines call groups for calls to this device (e.g. 1,3-5)

Pickup Group : Defines pickup groups from this device. (e.g. 1,3-5), calls can be picked up by dialing Group Pickup code (e.g. *8)   if this value matches the Call Group value of ringing extension.

Qualify : If checked, monitoring on network connection between server and user is enabled.

DTMF Mode : The DTMF standard to be used by this extension

● Automatic : Use rfc2833 for DTMF relay by default but will switch to inband DTMF tones if the remote side does not indicate support of rfc2833

● RFC2833 : Use RFC2833 for DTMF

● Inband :The device that you press the key on will generate the DTMF tones. If the codec is not ulaw or alaw then the DTMF tones will be distorted by the audio compression and will not be

recognised.

- Info   :Use SIP method INFO or SIP INFO as DTMF

Disallowed Codec   : Audio Codecs NOT to used by this extension

Use Snom Phone   : To specify if this extenion is connected by Snom phone or not, if enabled and with MAC address filled in, auto configuration for the IP Phone will be enabled

4. IAX2 Settings:

Group   : Dial plan group to be used

Host   : Allow Dynamic (any IP) host or Custom (fixed IP) host

Type   : Setup to allow Outgoing / Incoming / Both way calls

Overwrite CallerID   : Overwrite Caller-ID as User Name if checked

IAX Trunk   : Channel type that consume lower bandwidth. (Hardware timer support needed).

Can Transfer   : Allow transfer feature for this extension / account
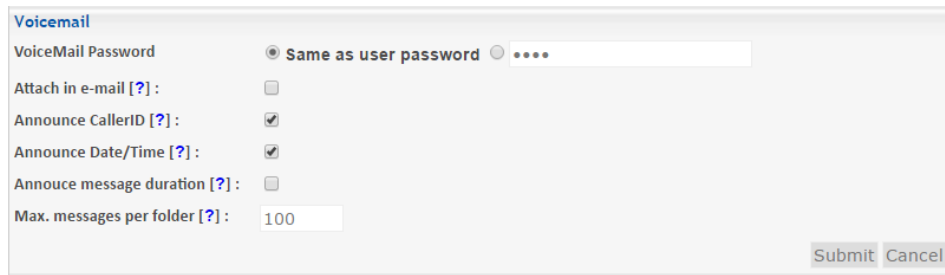
Qualify   : If checked, monitoring on network connection between server and user is enabled.

DTMF Mode   : The DTMF standard to be used by this extension

- Automatic : Use rfc2833 for DTMF relay by default but will switch to inband DTMF tones if the remote side does not indicate support of rfc2833

- RFC2833 : Use RFC2833 for DTMF

- Inband   :The device that you press the key on will generate the DTMF tones. If the codec is not ulaw or alaw then the DTMF tones will be distorted by the audio compression and will not be recognised.

- Info   :Use SIP method INFO or SIP INFO as DTMF

Disallowed Codec   : Audio Codecs NOT to used by this extension

5. Voice Mail Settings

VoiceMail Password   : Password for thie voice mailbox, can be same as User's Password setup in earlier secion, or specify a different password

Attach in e-mail   : Attach voice message in .wav format in e-mail if enabled (Requires E-mail Address setup in earlier section)

Annouce CallerID   : Announce Caller-ID for the caller before each message is played
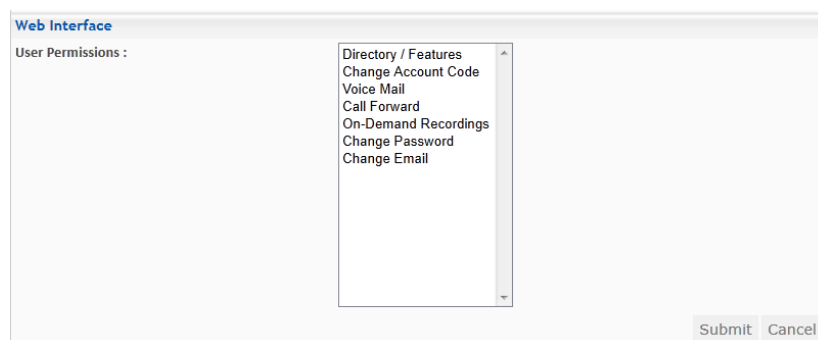
Announce Date/Time   : Announce Date and Time of the message left before each message is played

Annount Message Duration : Announce message duration of the message before each message is played

Max. messages per folder : Maximum number of message allowed to store in each folder

(Notice: if number of message has reached the maximum allowed limit, no more new message can be left on that mail box, all existing message will be kept)

6.   Web Interface Settings

Select the permissions for this user if web interface is enabled.

(Hints: Hold the "CTRL" key to select multiple modules)

7.   Press "Submit" button to complete

25

On Demand Recordings (Admin)

If On Demand Recording is activated during conversation, voice conversation will be logged and searchable from this module.

Retrieving On Demand Voice Logs

1. Input the search critirea such as Caller (called from), Destination (called to) and period of voice logs needed, please note that if no search critirea, system will show all call history.

2. The table shows the call details:

   Date   : Date and Time for the call made, in the format of YYYY-MM-DD HH:MM:SS

   Caller   : Telephone number that initiated the call, and in blanket showing the channel used

   Destination : Called telephone number, and in blanket showing the channel used

   Duration : Total talk time for the call (excluding Ringing time)

   Play   : Click to playback the voice log or save the voice log to computer

Voice Mail Management

Voice Mail Management module allows administrator to manage mailboxes within the system. Administrators may delete or listen to the voice mail within any extension's voice mail box.
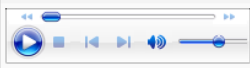
Retrieve Voice Messages

1. Click on the number within the "New/Inbox" column for new messages or the number within "Old" column for old messages.

2. Message List will be shown, information including Caller-ID, Date and Time, Duration for the message, click on any message listed to show message details

3. An embedded player will be available (currently supports Internet Explorer only), and details of the message will be shown:

Click on embedded player control keys to play back the voice message.

To download the voice message to computer, click on the link next to "Download"

Click on "Delete" to delete this message

Voice Mail Management (Cont')

Bulk Delete Messages

1. Click on Delete All within the "New/Inbox" column or "Old" column to delete all messages within the folder

Voice Mail Management					
Extension	New/Inbox		Old		Delete All ☐
101	0	(Delete All)	3	(Delete All)	☐
102	0	(Delete All)	0	(Delete All)	☐
106	3	(Delete All)	1	(Delete All)	☐
107	11	(Delete All)	0	(Delete All)	☐
108	0	(Delete All)	0	(Delete All)	☐
110	0	(Delete All)	0	(Delete All)	☐
111	4	(Delete All)	0	(Delete All)	☐
112	0	(Delete All)	0	(Delete All)	☐
113	3	(Delete All)	0	(Delete All)	☐
118	2	(Delete All)	0	(Delete All)	☐
					Delete

2. Check on several mail boxes to select multiple mail boxes for message removal, or even check on "Delete All" to select all mail boxes, press "Delete" button

Voice Mail Management					
Extension	New/Inbox		Old		Delete All ☐
101	0	(Delete All)	3	(Delete All)	☐
102	0	(Delete All)	0	(Delete All)	☐
106	3	(Delete All)	1	(Delete All)	☐
107	11	(Delete All)	0	(Delete All)	☑
108	0	(Delete All)	0	(Delete All)	☐
110	0	(Delete All)	0	(Delete All)	☐
111	4	(Delete All)	0	(Delete All)	☑
112	0	(Delete All)	0	(Delete All)	☐
113	3	(Delete All)	0	(Delete All)	☑
118	2	(Delete All)	0	(Delete All)	☑
					Delete

3. A list of delete list and confirm message will be shown, press "Yes" button to confirm or "No" button to cancel.

Voice Mail Management		
Extension	New/Inbox	Old
107	11	0
111	4	0
113	3	0
118	2	0
	Are you sure you want to delete this? Yes No	

Reset Mailbox Greetings

Personalized greeting can only be resetted by administrator in "Voice Mail management" module

1. Click on "Delete Greetings" next to the respective mailbox
2. Click "OK" when delete confirmation box is prompted

Appendix A: Platon Dial Plan Command List

Command	Variables Example(s)	Descriptions
Agent Login	-	To perform ACD Agent Login (for systems with ACD module only)
Agent Logoff	-	To perform ACD Agent Logout (for systems with ACD module only)
Answer	-	Answer an incoming call by the PBX, some dial plan commands requires the PBX to answer the call first before it works
Ask for passcode	1357	To prompt for input passcode before running the rest commands
Change CallerID (Name)	Example 1: Peter Chan Example 2: HK-${CALLERID(name)}	Any text or numbers to replace the original Caller Name, or put in front or after ${CALLERID(name)} to add or append a caller ID Name, for example: Original Caller Name is: Peter Variable input: HK-${CALLERID(name)} New Caller Name becomes: HK-Peter
Change CallerID (Number)	Example 1: 12345678 Example 2: 9${CALLERID(num)}	Any numbers to replace the original Caller Number, or put in front or after ${CALLERID(num)} to add or append a caller ID Number, for example: Original Caller Name is: 12345678 Variable input: 9${CALLERID(num)} New Caller number becomes: 912345678
Change Language	Cantonese	Choose any langauge from drop down list, or press "Edit" to add/modify existing language. Changing language will affect the system to apply different language files for voice play backs afterwards
Change Priority	High(5)	Change the caller's priority if it routed into Queue (For systems with ACD module and Membership System only)
Customized Command	-	Reservered for Platon Engineers and special features only
Customized Macro	-	Reservered for Platon Engineers and special features only
Dial by CO Trunk / Dial by IAX Trunk / Dial by SIP Trunk	CO (from first line) Prefix: 133 Strip Digits: 1 Timeout(in seconds): 15	Pass the dialed numbers to specifed Trunk, with the assigned line usage method (e.g. from first line). If prefix is specifed, it will be added in front of the dialed numbers and send to the specifed trunk. E.g. original number 123456, with Prefix 133, system will send 133123456 If strip digits is specified, system will remove the number of

		digits specified from the front of the dialed numbers
		E.g. original number 123456, with strip digits 1, system will send 23456
		Example for both prefix and strip digits were set:
		Original Number: 123456, prefix 133, strip digits 1
		System will send out: 13323456
		Timeout sets the time for this dial command to be run, if set, it will ring for the trunk for specifed time (in seconds)
Dial Extensions	Peter Chan (101) Timeout (in seconds): 15	Select one or multiple extension(s) to ring, if timeout value have been set, the selected extension(s) will be rang for specified time only.
Dial Extension (according to dial rule)	Prefix: 2 Strip Digits: 1	Dial to an extension acoording to dial rule. If prefix is specifed, it will be added in front of the dialed numbers and send to the specifed trunk. E.g. original number 201, with Prefix 2, system will send 2201 If strip digits is specified, system will remove the number of digits specified from the front of the dialed numbers E.g. original number 201, with strip digits 1, system will send 01
Dial Extension Group	Admin Department	Dial to the specified extension group Extension group members can be setup in Extension Group module
Dial Phone No.	SIP/192.168.0.1	Dial to specifed phone number Works with any VOIP URI
Direct Pickup	Prefix: 2 Strip Digits: 1	To perform extension direct pickup.
DISA	Password: 1357 Dialplan Group: default	DISA (Direct Inward System Access) To provide internal dial features by calling into the system Password field specifies the password to run DISA Dialplan Group to specify the dialplan used when using DISA
Fax Extension	611 (Peter Chan)	Redirect the call to the specified fax box (for systems with Fax module only)
Fax reception	Prefix: 2 Strip Digits: 1	Redirect the call to fax boxes according to the dial rule
Hangup	-	To hangup the call, any commands followed by this command will not be handled.
IVR Menu	Main IVR	Redirect the call to specified IVR Menu.

		IVR Menu can be managed in IVR Menu module
Jump to another DialRule	Default: *01 Priority: 1	Redirect the call to another dial rule and start from the specified command number
Listen to Voicemail	-	Goto Voice Mail Management system, allows user to listen to voice mail, mangement greetings and passwords. If "Read CallerID as Extension" is checked, CallerID will be used as Login ID for the voice mail system
Mailbox	Andy Lee (102) Busy	Redirects the call to specified mailbox and play the specifed prompt. Busy : Play Busy greeting message Unavailable : Play Unavailable greeting message No Prompt : Play a beep only
Mailbox (according to dial rule)	Prefix: 2 Strip Digits: 1 Unavailable	Redirects the call to mailbox according to dialed number and play the specifed prompt. Busy : Play Busy greeting message Unavailable : Play Unavailable greeting message No Prompt : Play a beep only
MeetMe Conference	-	MeetMe Conference / Conference Bridge First user create pin: First user creates the conference pin number; Announce user count on joining: System will announce number of users joined; Announce user join/leave: System will announce the name of user that joined or leave the conference, each user will be required to record their name before they join the conference Record the conference: Conference conversations will be recorded
Membership Validation	Play: pls-ent-membership-no Time-out: 5 seconds Max. Digits (Optional): 6 If valid, go to step: 2 If invalid, go to step: 7 If timeout, play: Please try again	To enable Membership validation (For systems with membership validation add-ons only) Membership ID can be validated by this command, the system will compare with the membership table imported earlier and goes to appropriate dial command steps for members and non-members Membership ID will be saved as ${MEMID} variable within the call. (Which can be displayed as Caller Name)
Page Extension	Prefix: 2 Strip Digits: 1	Page the extension (Intercom) according to the number dialed
Page Extension Group	Admin Department	Page the specified extension group

		Extension group members can be setup in Extension Group module
Page Extensions	Peter Chan (101)	Select one or multiple extension(s) to page/intercom
Play Busy Tone	for 5 seconds	Play busy tone to the caller for specifed time
Play Congestion Tone	-	Play congestion tone to the caller
Play Sound	Thank-you-for-calling	Play the specifed voice (If used in IVR, no key input will be detected)
Play Sound (Background)	Thank-you-for-calling	Play the specifed voice with digit input accepted This command can be used within IVR only
Prompt for Account Code	-	To prompt for account code and wait for correct account code before proceeding commands followed by this. Usage Example: IDD dialing Account Codes can be managed be Account Codes Management module
Queue	Customer Service (cs) Timeout: 45 seconds	Redirect calls into specified ACD Queue (for systems with ACD module only) If timout is set, calls will be hold in queue for the time specified and run the next command if timeout
Realtime Monitor	SIP Channels	To perform real-time call monitoring on specified channels. A random channel/extension will be picked for call listening, press * to pick the next extension
Redirect to another DialRule	Prefix: 2 Strip Digits: 1 Dialgroup: default	To redirect the call to similar dial rule in another dialgroup Example: Original Dial Rule: 150 With prefix: 2, strip digits 1, dialgroup default Calls will be redirected to 250 in default dialgroup
Run Hangup Script	-	Reserved for Platon Engineers and special features only
Run Script	-	Reserved for Platon Engineers and special features only
Show on CLI	-	Text to be shown on console Reserved for Platon Engineers and special features only
Wait for Input	for 8 seconds	Used wihtin IVR Menu, to wait for digit input for the specified time
Wait/Delay	For 5 seconds	To make a delay for the specified time before running the next command. Usually used after Answer command, for better compatibility with slow responsed systems