Avaya J100 Series IP Phones

Partner configuration guide for FreeSWITCH

April 2019

Document Version 1.0

Contents

Overview

This guide describes the configuration procedures required for the Avaya J100 Series IP Phone for interoperability with FreeSWITCH. This includes the following models.

- J129
- J139
- J169
- J179

The J100 is a desktop phone that uses the Session Initiation Protocol (SIP) to communicate with FreeSWITCH for call control.

This guide describes the specific configuration items those are important for use with FreeSWITCH. It does not describe the purpose and use of all configuration items on the J100. For those details, see the *Installing and Administering Avaya J100 IP Phone in third-party call control setup* [1]

Interoperability Status

This section provides the known interoperability status of the Avaya J100 IP Phone with FreeSWITCH. This includes the version(s) tested, the capabilities supported, and known issues.

Verified Versions

The following table identifies the verified Avaya J100 IP Phone and FreeSWITCH versions and the month/year the testing occurred.

Verifications			
Date (mm/yyyy)	**FreeSWITCH Release**	**J100 Verified Version**	**J100 Compatible Versions**
04/2019	1.8.5	4.0.1.0	Any maintenance release of validated version.

Known Issues

This section lists the known interoperability issues between FreeSWITCH and J100 release 4.0.1.0.

The following table provides a description of each issue and, where possible, identifies a workaround.

Issue Number	**Issue Description**
SIP96X1-52487:	The voice is bad quality when using codec OPUS
SIP96X1-54680	Phone receives a BLF incoming call when the monitored phone holds
SIP96X1-54762	Phone display caller ID after Hold/Resume in call privacy
SIP96X1-54791	Call log displays "caller ID" instead of "Restricted" in privacy call if phone activates call forward
SIP96X1-56316	Phone cannot detect the state of BLF phone if SIP domain and SIP proxy list are different

FreeSWITCH Configuration

J100 IP Phone Configuration

This section describes the configuration settings required for the J100 integration with FreeSWITCH, primarily focusing on the SIP interface configuration. The J100 configuration settings identified in this section have been derived and verified through interoperability testing with FreeSWITCH. For configuration details not covered in this section, see the *Installing and Administering Avaya J100 IP Phone in Third-Party Call Control Setup* [1] for J100.

Configuration Method

Avaya J100 IP Phone can be configured using the 46xxsettings file. The phone can access the settings file via HTTP and HTTPS.

Configuration Files

Avaya J100 Configuration Files	Level	Description
J100Supgrade.txt	System	Contains the device firmware load.
46xxsettings.txt	System	Contains configurable parameters that apply to all devices in a given deployment.
<MACaddress>.txt	Subscriber	Contains configurable parameters that apply to an individual device in a deployment.

System Level Configuration

This section describes system-wide configuration items in the 46xxsettings.txt file that are generally required for each Avaya J100 IP Phone to work with FreeSWITCH. Subscriber-specific settings are described in the next section.

Step 1	**`SET ENABLE_AVAYA_ENVIRONMENT 0`**
Step 2	`SET DISCOVER_AVAYA_ENVIRONMENT 0`
Step 3	`SET ENABLE_IPOFFICE 0`
Step 4	`SET ENABLE_3PCC_ENVIRONMENT 1`
Step 5	`SET 3PCC_SERVER_MODE 0`

Configure Network Settings

Step	Command
Step 1	SET ENABLE_UDP_TRANSPORT 1
Step 2	SET DNSSRVR “8.8.8.8”
Step 3	SET DOMAIN “”
Step 4	SET SNTPSRVR pool.ntp.org
Step 5	SET SNTP_SYNC_INTERVAL 144000

Configure SIP Interface Settings

Step	Command
Step 1	SET SIPDOMAIN "FreeSWITCH.net"
Step 2	SET SIP_CONTROLLER_LIST "FreeSWITCH.net:5060;transport=udp" OR SET SIP_CONTROLLER_LIST "10.133.46.166:5060;transport=udp"
Step 3	SET ENABLE_G711A 1
Step 4	SET ENABLE_G711U 1
Step 5	SET ENABLE_G722 1
Step 6	SET ENABLE_G726 0
Step 7	SET G726_PAYLOAD_TYPE 110
Step 8	SET ENABLE_G729 1
Step 9	SET ENABLE_OPUS 0
Step 10	SET SEND_DTMF_TYPE 2
Step 11	SET DTMF_PAYLOAD_TYPE 120
Step 12	SET 100REL_SUPPORT 1
Step 13	SET PLAY_TONE_UNTIL_RTP 1
Step 14	SET SYMMETRIC_RTP 1
Step 15	SET REGISTERWAIT 1200
Step 16	SET WAIT_FOR_UNREGISTRATION_TIMER 32
Step 17	SET WAIT_FOR_INVITE_RESPONSE_TIMEOUT 60
Step 18	SET FAILED_SESSION_REMOVAL_TIMER 30
Step 19	SET TCP_KEEP_ALIVE_STATUS 1
Step 20	SET TCP_KEEP_ALIVE_TIME 60
Step 21	SET TCP_KEEP_ALIVE_INTERVAL 10
Step 22	SET SIP_TIMER_T1 500

Step	Command
Step 23	SET SIP_TIMER_T2 4000
Step 24	SET SIP_TIMER_T4 5000
Step 25	SET ENABLE_SIP_USER_ID 1
Step 26	SET SIMULTANEOUS_REGISTRATIONS 1
Step 27	SET LOCALLY_ENFORCE_PRIVACY_HEADER 1
Step 28	SET ENABLE_STRICT_USER_VALIDATION 0

Configure Service Settings

Step	Command
Step 1	SET DIALPLAN [23]xxxx|91xxxxxxxxxx|9[2-9]xxxxxxxxx|7xxx
Step 2	SET NO_DIGITS_TIMEOUT 20
Step 3	SET INTER_DIGIT_TIMEOUT 5
Step 4	SET ENABLE_DND_PRIORITY_OVER_CFU_CFB 0
Step 5	SET ENABLE_AUTO_ANSWER_SUPPORT 1
Step 6	SET AUTO_ANSWER_MUTE_ENABLE 1
Step 7	SET HOLD_REMINDER_TIMER 0
Step 8	SET CONFERENCE_FACTORY_URI "nway@10.133.46.166 "
Step 9	SET SIPCONFERENCECONTINUE 0
Step 10	SET PSTN_VM_NUM "*"
Step 11	SET SUBSCRIBE_LIST_NON_AVAYA "message-summary"
Step 12	SET RINGTONESTYLE 0
Step 13	SET 3PCC_SERVER_MODE 0

Subscriber Level Configuration

This section identifies the device-specific parameters, including registration and authentication. These settings must be unique across devices in order to be matched with the settings for a FreeSWITCH SIP trunk or subscriber. .

Step	Command
Step 1	SET FORCE_SIP_USERNAME "1001"
Step 2	SET FORCE_SIP_PASSWORD "123456"
Step 3	SET FORCE_SIP_EXTENSION "1001"
Step 4	SET COUNTRY USA
Step 5	SET SYSTEM_LANGUAGE Mlf_J100_English.xml

Step 6	SET LANGUAGES "Mlf_J100_CanadianFrench.xml,Mlf_J100_LatinAmericanSpanish.xml,Mlf_J100_German.xml"
Step 7	SET DAYLIGHT_SAVING_SETTING_MODE 2
Step 8	SET DSTOFFSET 1
Step 9	SET DSTSTART 2SunMar2L
Step 10	SET DSTSTOP 1SunNov2L
Step 11	SET GMTOFFSET 0:00

SIP Advanced Feature Configuration

Busy Lamp Field Configuration

BLFs are used to display state of monitored users and can be used as hotkey for specific operations:

- Speed dial to monitored user.
- Pick up an incoming call.
- Barge in an active call.
- Transfer and conference using BLF.

- In settings file, add below command to configure BLF.

  SET PHONEKEY "key=25;type=feature;name=blf;attr1=2301;Label=BLF2301"

- Transfer and conference using BLF:

  - Transfer: While phone is in active call, pressing Transfer SK after pressing BLF line key, phone will transfer the call to BLF phone.

  - Conference: While phone is in active call and pressing Conf SK after pressing BLF line key, phone will add BLF phone to a conference.

Call Transfer

Call transfer on FreeSwitch supports three types of transfer as below:

- Blind transfer.

- Attended transfer.
- Un-attended transfer.

Note: Phone in transfer call still displays transferred user's info on active line.

For example: phone A and phone B are in active call. Phone A transfers successfully the call to phone C but phone B still displays info of user A.

Conference

1. Local Conference

Local conference works well on J100 series

2. Ad-hoc Conference

2.1. Configuration

2.2.1.1. Configuration on FusionPBX

On Fusion PBX, go to **Dialplan** > **Dialplan Manager** > set **true** for '**nway_conference**' field to enable Ad-hoc conference.

2.2.1.2. Configuration in settings file

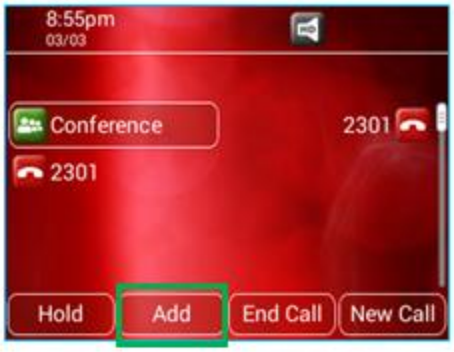
Add below command in setting file

SET CONFERENCE_FACTORY_URI nway@10.16.25.95

2.2. Create Ad-hoc conference

To create an ad-hoc conference, we follow the steps below:

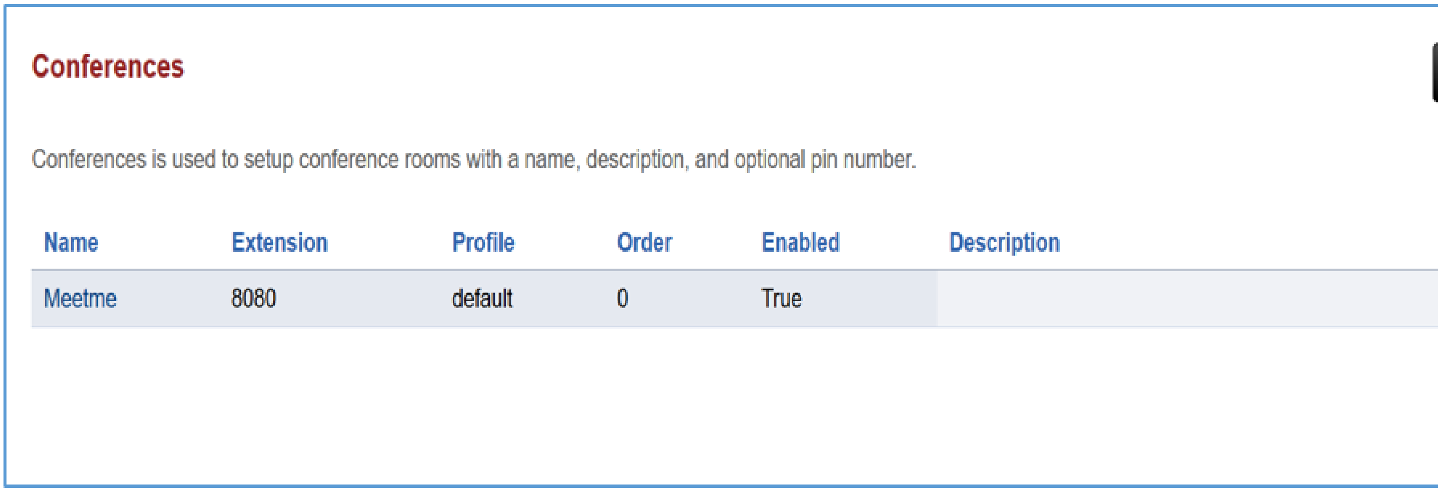
- Phone A makes a call to phone B and phone B answer this call.
- On phone A, press **Conf** softkey and dial to phone C to create a conference. There is a conference between phone A, B and C.
- On phone A, press **Add** sofkey to add more user into the conference.



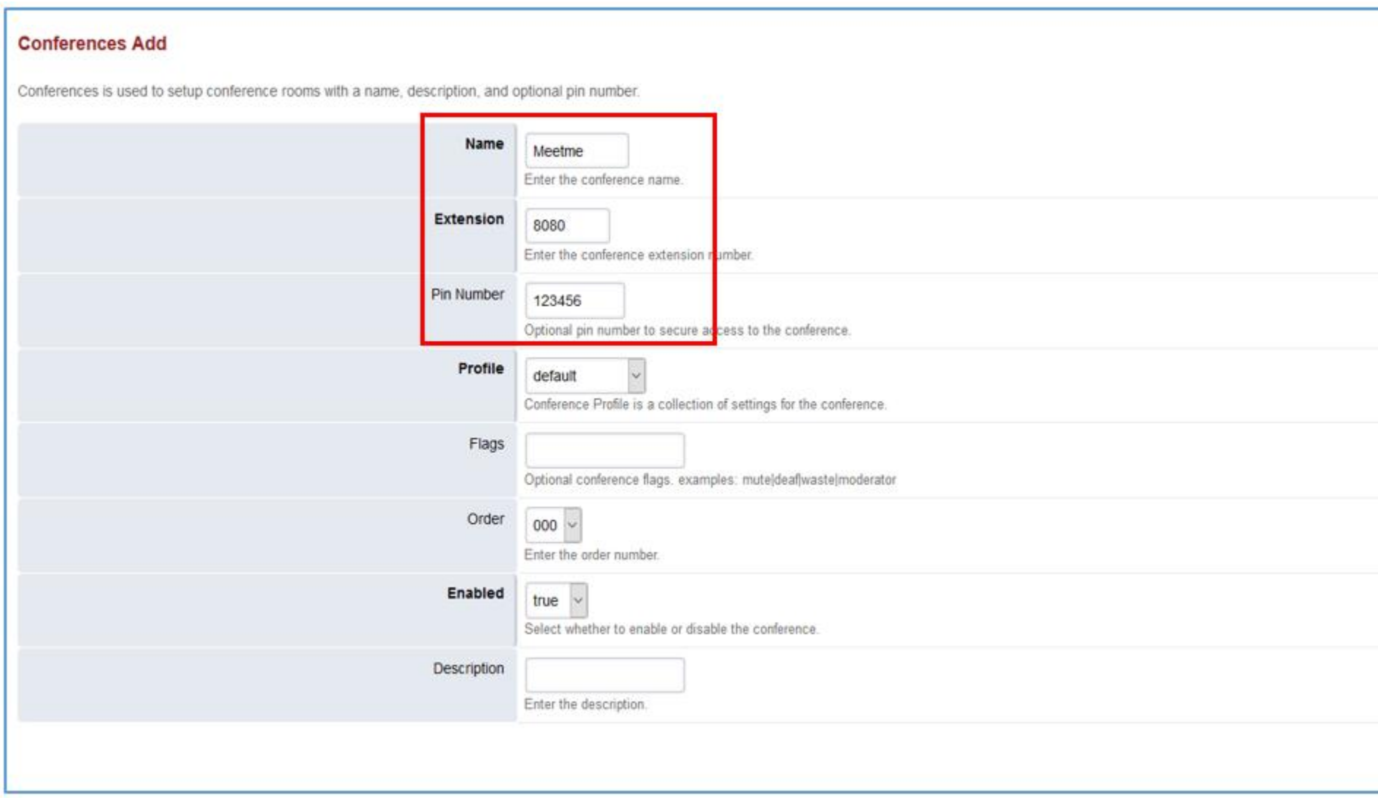
2.3. Meet-me conference

2.3.1. Configuration

On FusionPBX, go to Apps > Conferences > select the plus icon as below to create meet-me conference number.



Fill the required information and select save to complete



2.3.2. Create a meet-me conference

To create a meet-me conference, we follow the steps below:

- Dial 8080 (this is a meet-me conference number which is configured above).
- Enter PIN number to create a meet-me conference.
- On other phone, dial 8080 and enter PIN number to join the meet-me conference.

Voice Message

4.4.4.1 Configuration

Configuration voice message on FusionPBX:

- To edit voicemail settings: Go to Apps > Voicemail > Click the pencil icon

Voicemails (22)

Voicemail settings.

Voicemail ID	Mail To	Attached	Keep Local	Tools	Enabled	Description
1001		True	True	Messages Greetings	True	
1002		True	True	Messages Greetings	True	
1003		True	True	Messages Greetings	True	
1004		True	True	Messages Greetings	True	
1005		True	True	Messages Greetings	True	
2004		True	True	Messages Greetings	True	
2005		True	True	Messages Greetings	True	
2006		True	True	Messages Greetings	True	
2007		True	True	Messages Greetings	True	
2300		True	True	Messages Greetings	False	
2301		True	True	Messages Greetings	True	
2302		True	True	Messages Greetings	True	
2303		True	True	Messages Greetings	True	
2304		True	True	Messages Greetings	True	
2305		True	True	Messages Greetings	True	

4.4.4.2 Voicemail options

FACs for voicemail as below:

*97	To access that extensions voicemail from the extension or the voicemail button
*98	To access any extensions voicemail
*99[ext]	To access a specific extension voicemail

4.4.4.3 Send and receive voice message

4.4.4.3.1 Send voice message

We send voice message follows 2 ways as below:

- Dial the extension and leave a voice message if this extension does not answer this call
- Dial ***99[extension]** and leave a voice message to extension

4.4.4.3.2 Receive voice message

- From the extension has new voicemail: Dial ***97** and enter correct voicemail password to access voice massage.
- From any extensions: Dial ***98** and enter extension, voice mail password to access voicemail.

Call Forward

4.4.5.1. Local call forward

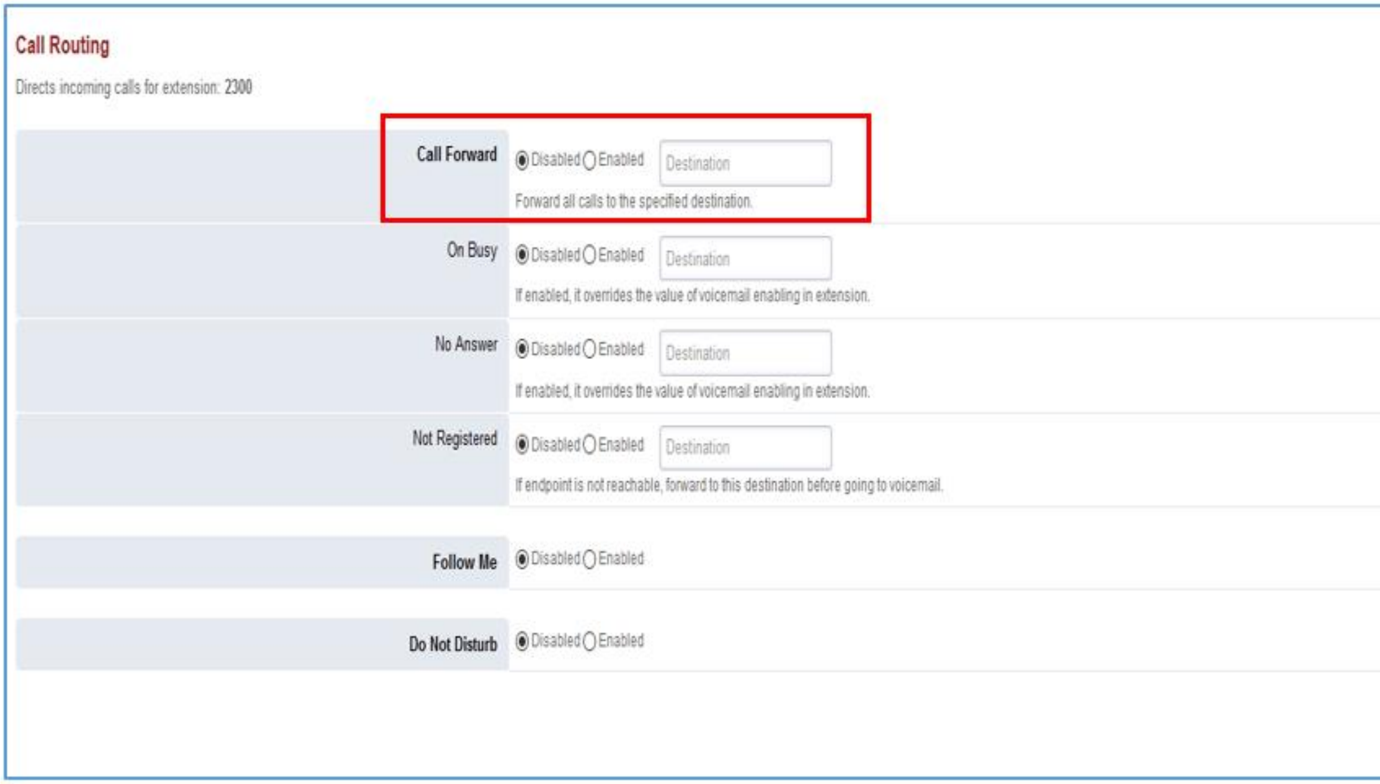
- Add **SET CALLFWDSTAT 7 in the settings file** to enable call forward all, busy and no answer feature
- On phone, go to **Feature** à Select Call forward and set the destination extension to enable Call forward

4.4.5.2 System Call forward

Call forward – All

There are 2 ways to setup Call forward – All:

- On system, go to Apps > Call Routing > select the extension à Enable Call forward and set the destination extension



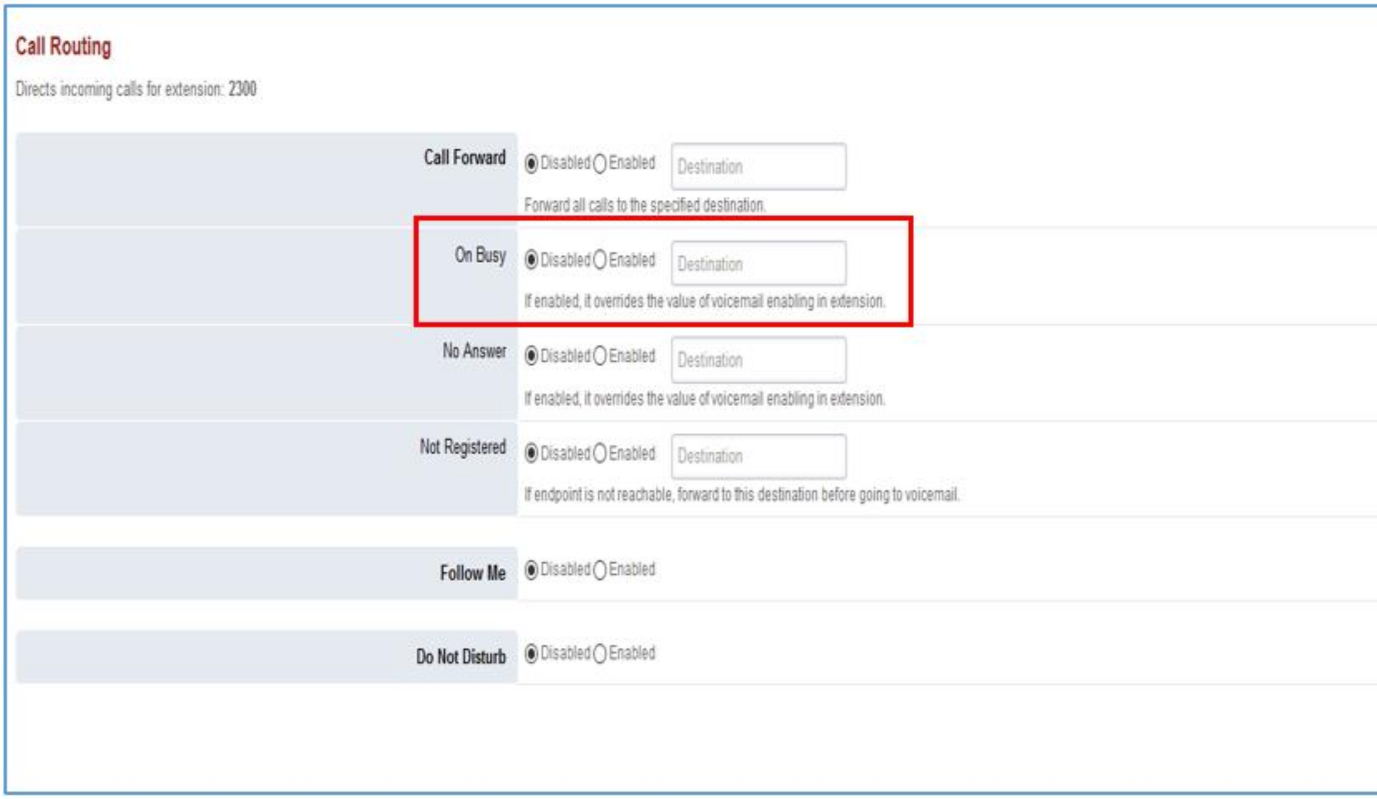
- Using FAC:

*72	Enable Call Forward
*73	Disable CallForward

Dial *72 > Enter the destination extension to enable Call forward - all

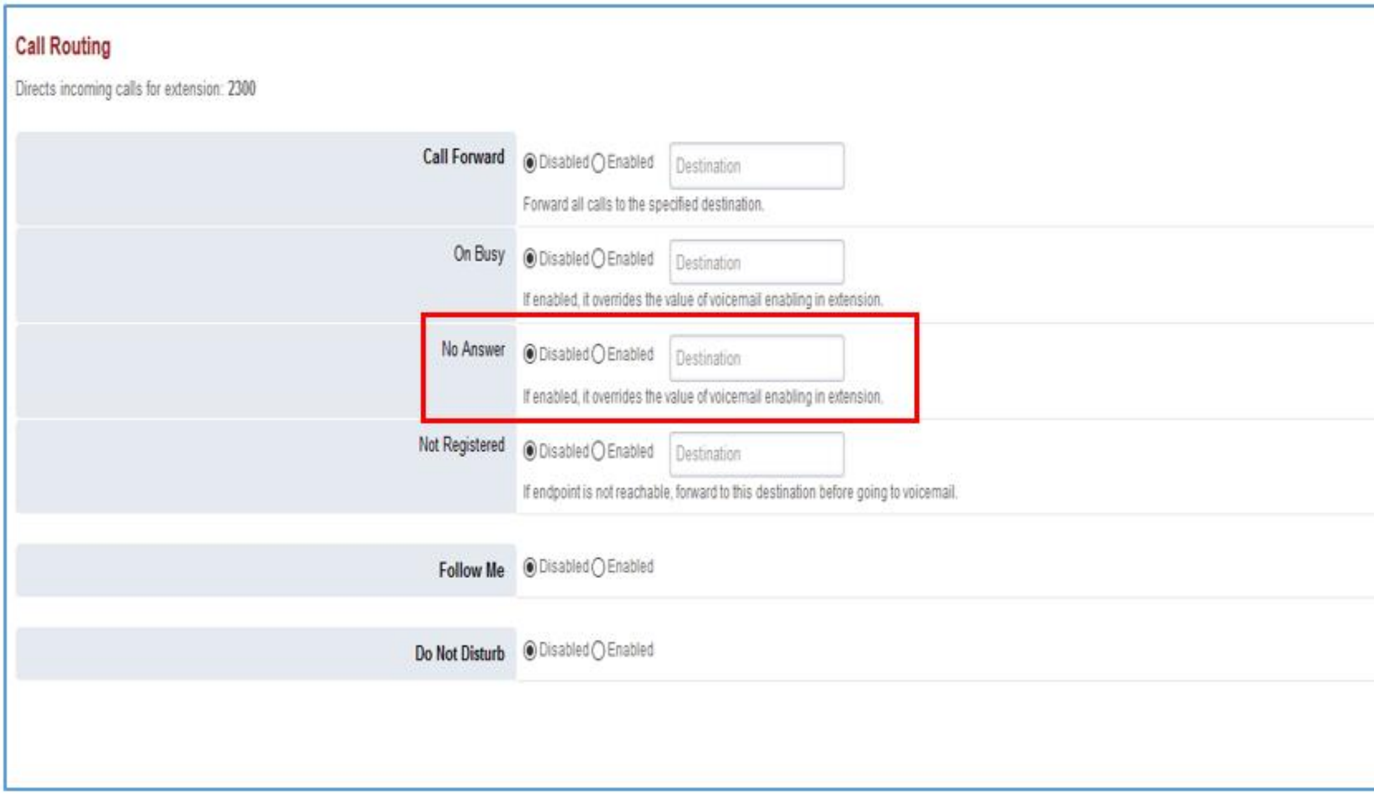
Call forward – Busy

On system, go to Apps > Call Routing > select the extension > Enable Call forward on Busy and set the destination extension



Call forward – No answer

On system, go to Apps > Call Routing > select the extension > Enable Call forward No answer and set the destination extension



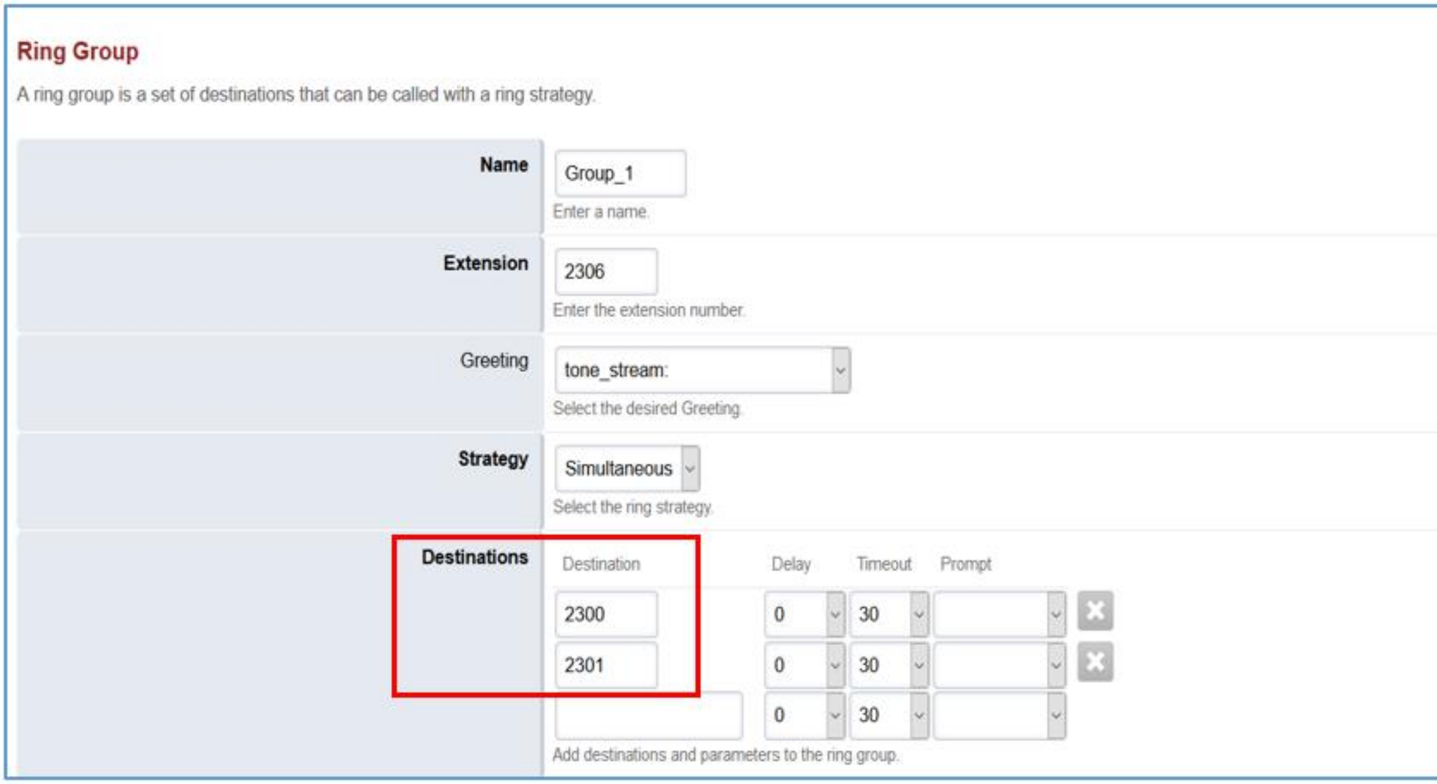
Call Pickup

Configuration:

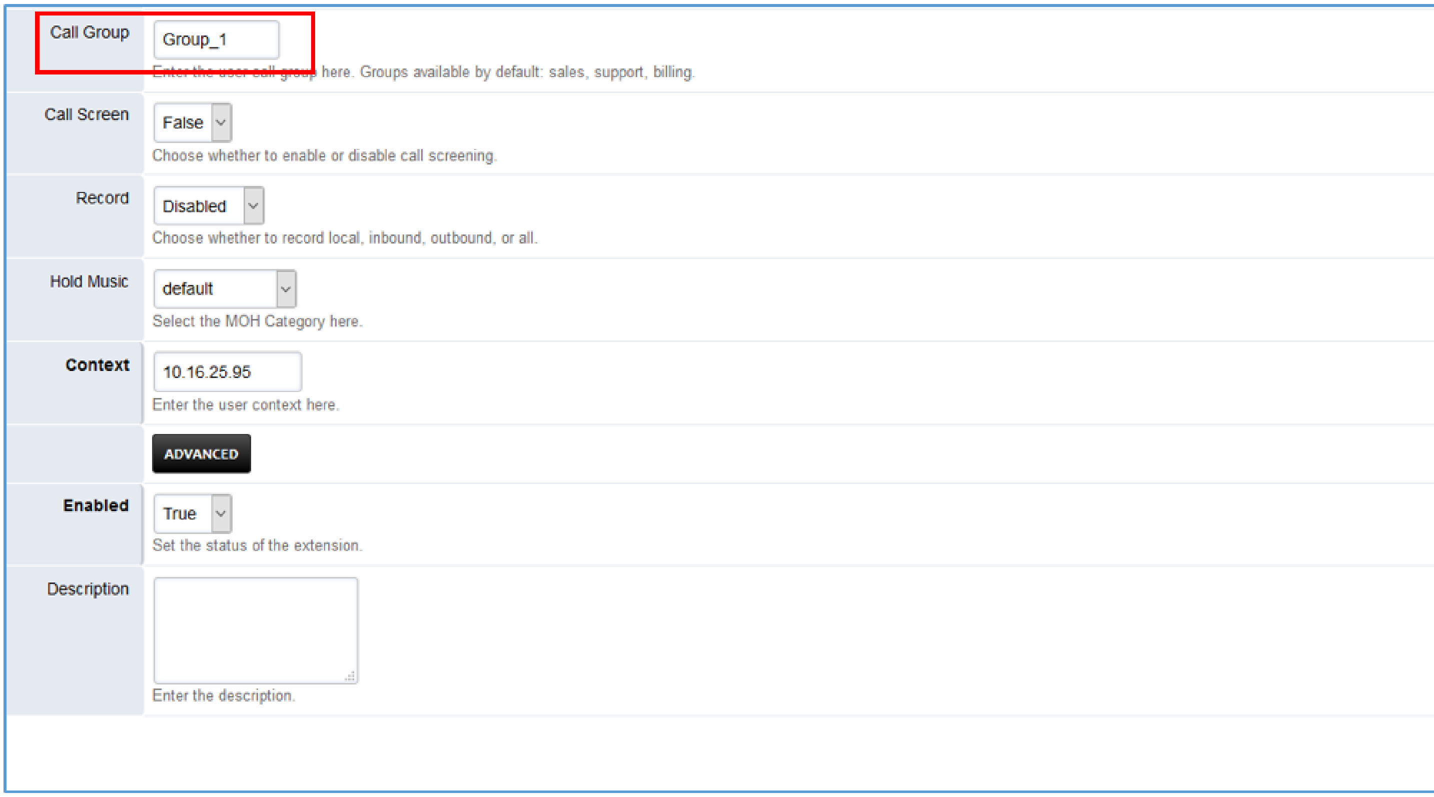
- Create ring group:

  - On system, go to Apps > Ring Groups > Select the plus icon as figure below:

- In this example, we will have 2 extensions (2300, 2301) all ring at the same time until one of them pick up the call.



- Add the extensions into Ring Group: On system, go to **Extension** > select the extensions which you want to add into ring group > Enter group name (Group_1) into **Call Group** field and press **Save**



Pickup a call

Pick up a call in the same group

Users in the same group can dial ***8** to answer the incoming call for any members of ring group.

For example: Phone A and B are in a ring group. Phone A has an incoming call from phone C, phone B dials ***8** to pick up this call.

Directed Call pickup

With Directed call pickup, users can pick up an incoming call of any users using FAC: ****[extension]**. A Ring group is not required with Directed Call pickup.

Call Park

- To view FAC for call park: Go to Dialplan > Dialplan Manager > valet_park: *5900 - *5999
- User can park the call by doing Blind Transfer to *5900 - *5999 and other users dial *5900 - *5999 to pick up the call. The user will park and retrieve the call from the same plot.

For example: User A and B are in an active call. user A do Blind transfer and dial *5900 to park the call. Phone C will dial *5900 to retrieve this call

Do Not Disturb

Local DND

In settings file, add **SET ENABLE_DND 1** command to enable DND feature. Go to Feature on phone and select DND to activate.

System DND

- On system, go to Apps > Call routing > Extension > select **Enabled** and save to active DND

- Using FAC to activate/ deactivate DND follows table below:

*78	Enable Do Not Disturb

*79	Disable Do Not Disturb

Note:

- All calls will be forwarded to Voice mail in case of enabled DND.
- If DND and Call forward both are enabled on a phone, Call forward will takes priority.

Emergency Call

- In settings file, add the commands as below:

SET PHNEMERGNUM 9911

SET ENABLE_SHOW_EMERG_SK 2

Note:

- Emergency call can be made using Emerg soft key and on-hook dialing
- Call initiator won't be able to Drop or Hold the emergency call.

Anonymous Call

- In settings file, add command: SET LOCALLY_ENFORCE_PRIVACY_HEADER 1
- Dial ***67[extension]** to make an Anonymous Call. The phone receives an Anonymous Call and it will be displayed as "**Restricted**" as figure below:

Note: Call log for an anonymous call displays as Restricted.

Appendix A: Reference J100 Configuration Files

The following is a reference configuration for the J100 configured for use with FreeSWITCH.

System Default File: 46xxsettings.txt

NOTE: This is an example file and it should be used for reference only.

```
##################################################################################
##################################
## ENABLE_AVAYA_ENVIRONMENT specifies whether the telephone is configured
##  for use in an Avaya (SES) or a third-party proxy environment.
##  Value  Operation
##    0    3rd party proxy with "SIPPING 19" features
##    1    Avaya SES with AST features and PPM (default)
##################################################################################
##################################

SET ENABLE_AVAYA_ENVIRONMENT 0

##################################################################################
##################################
##  If the DISCOVER_AVAYA_ENVIRONMENT parameter value is 1, the phone discovers
(determines)
##  if that controller supports the AST feature set or not. The phone will send a
SUBSCRIBE
##  request to the active controller for the Feature Status Event Package (avaya-
cm-feature-status).
##  If the request succeeds, then the phone proceeds with PPM Synchronization.
##  If the request is rejected, is proxied back to the phone or does not receive
a response,
##  the phone will assume that AST features are not available.
##  If the parameter value is 0, the phone operates in a mode where AST features
are not available.
##  Note: This parameter is supported on J129  SIP  R1.0.0.0 (or R1.1.0.0) and
J100 SIP R2.0.0.0 and later.
##  For IP office and 3PCC environments this parameter shall be set to 0.
##################################################################################
##################################

SET DISCOVER_AVAYA_ENVIRONMENT 0

##################################################################################
##################################
## ENABLE_IPOFFICE specifies whether the deployment environment is IP Office
##  Value  Operation
##   0        Not IP Office environment (except failover mode to IP Office in
Avaya Aura environment) (Default)
##   1     IP Office environment; Native support of IP Office with a limited
feature set.
##   2        IP Office environment; Additional features driven by the IP Office
SIP proxy.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J129 supports values 0-1 and J169/J179
support values 0 and 2). When ENABLE_IPOFFICE is set to 2,
##           some of the 46xxsettings.txt file parameters have no effect.
```

```
##       Avaya Vantage Basic Application SIP R1.1.0.1 and later (values 0-1)
##       Avaya Vantage Devices SIP R1.1.0.1 and later (values 0-1)
##       J169/J179 SIP R1.5.0 (values 0-1)
##       J129 SIP R1.0.0.0 (or R1.1.0.0) (values 0-1)
##################################################################################
##################################

SET ENABLE_IPOFFICE 0

##################################################################################
##################################
## ENABLE_3PCC_ENVIRONMENT specifies whether the deployment environment is third
party SIP Server
##  Value  Operation
##   0        Not 3PCC environment
##   1     3PCC environment (Default)
##  Note: This parameter should be set to '0' for Aura environment and IP Office
##  Note: This parameter is supported by J129  SIP  R1.1.0.0 and J100 SIP
R2.0.0.0 and later
##################################################################################
##################################

SET ENABLE_3PCC_ENVIRONMENT  1

##################################################################################
##################################
## ENABLE_STRICT_USER_VALIDATION specifies whether AOR received in 'Request-URI'
of incoming call should be validated
## or not with 'contact' header published by phone in REGISTRATION.
##  Value  Operation
##   0        validation is not done (Default)
##   1     validation is done
##  Note: This parameter is supported by J129 SIP R1.1.0.0 and J100 SIP R2.0.0.0
and later only for 3PCC environment.
##################################################################################
##################################

SET ENABLE_STRICT_USER_VALIDATION  0


##################################################################################
##################################
## ADMIN_PASSWORD specifies a complex access code for access to local (craft)
procedures.
##  Valid values contain 6 and 31 alphanumeric characters including upper, lower
and special characters.
##  The default value is 27238 which implies that PROCPSWD is used as access code
for access to local (craft) procedures.
##  If ADMIN_PASSWORD length is less than 6 or greater than 31, the parameter is
treated as not defined.
##  If ADMIN_PASSWORD is configured, then PROCPSWD is ignored.
##  The special characters supported are: ~!@#$%^&*_-+=`|\(){}[]:;'<>,.?/. " is
not supported.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP
R2.0.0.0 and later
## eg: SET ADMIN_PASSWORD 123456
## use TAG
##################################################################################
##################################
```

```
SET ADMIN_PASSWORD 123456

####################################################################################
##################################
## PROCPSWD specifies an access code for access to local (craft) procedures.
##  Valid values contain 0 through 7 ASCII numeric digits.
##  The default value is 27238 (CRAFT) unless indicated otherwise below.
##  A null value implies that an access code is not required for access.
##  Note: Setting this parameter via CM (for H.323) or PPM (for SIP) is more
secure
##        because this file can usually be accessed and read by anyone on the
network.
##        Setting the value in this file is intended primarily for configurations
with
##        versions of telephone or server software that do not support setting
this
##        value from the server.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP
R2.0.0.0 and later (must contain at least 4 digits
##                          else default value 27238 is used)
## eg: SET PROCPSWD 123456
####################################################################################
##################################

SET PROCPSWD 27238

####################################################################################
##################################
## ADMIN_LOGIN_ATTEMPT_ALLOWED specifies the number of failed attempts for
entering the access code (PROCPSWD or ADMIN_PASSWORD)
## before the local (craft) procedures will be locked for a period specified by
ADMIN_LOGIN_LOCKED_TIME.
## Valid values are 1 to 20, default 10.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP
R2.0.0.0 and later
## SET ADMIN_LOGIN_ATTEMPT_ALLOWED 2
####################################################################################
##################################

SET ADMIN_LOGIN_ATTEMPT_ALLOWED 2

####################################################################################
##################################
## ADMIN_LOGIN_LOCKED_TIME specifies the time in minutes that local (craft)
procedures are locked once the
## number of failed attempts for entering the access code (PROCPSWD or
ADMIN_PASSWORD) is reached.
## Valid values 5 min to 1440 min, default 10 min.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP
R2.0.0.0 and later
## SET ADMIN_LOGIN_LOCKED_TIME 5
####################################################################################
##################################

SET ADMIN_LOGIN_LOCKED_TIME 5
```

```
###################################################################################
##################################
## SIP_CONTROLLER_LIST specifies a list of SIP controller designators,
##  separated by commas without any intervening spaces,
##  where each controller designator has the following format:
##  host[:port][;transport=xxx]
##  host is an IP address in dotted-decimal (DNS name format is not supported
unless stated otherwise below).
##  [:port] is an optional port number.
##  [;transport=xxx] is an optional transport type where xxx can be tls, tcp, or
udp.
##  If a port number is not specified a default value of 5060 for TCP and UDP or
5061 for TLS is used.
##  If a transport type is not specified, a default value of tls is used.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0); J100 SIP R2.0.0.0 and later; DNS name
format is supported for 3PCC environment only.
##      For 3PCC environment, only one SIP controller is supported.
##   J169/J179 SIP R1.5.0, DNS name format is supported.
## SET SIP_CONTROLLER_LIST "as.iop1.FreeSWITCH.net:5060;transport=tcp"
## Use Broad work TAG and custome TAG
###################################################################################
##################################

##SET SIP_CONTROLLER_LIST "10.133.46.166:5060;transport=udp"


###################################################################################
##################################
## ENABLE_SIP_USER_ID controls the display of the user ID input field on the
Login Screen
##  Value  Operation
##    0    SIP User ID field is not available to user during Login default)
##    1    SIP User ID field is available to user during Login
##  Note: This parameter is supported by J129 SIP R1.1.0.0 and J100 SIP R2.0.0.0
and later only for 3PCC environment
## SET ENABLE_SIP_USER_ID 1
###################################################################################
##################################

SET ENABLE_SIP_USER_ID 1

###################################################################################
##################################
## SIP Transport UDP
## Determines whether SIP Transport = UDP can be manually configured on the
phone.
##     0 for No (default)
##     1 for Yes
##  Note: This parameter is supported by J129 SIP R1.1.0.0 and J100 SIP R2.0.0.0
and later only for 3PCC environment.
## SET ENABLE_UDP_TRANSPORT 1
###################################################################################
##################################

SET ENABLE_UDP_TRANSPORT 1
```

```
###################################################################################
##################################
## Telephone number to call into the messaging system
##  PSTN_VM_NUM is the "dialable" string is used to call into the messaging
system
##  (e.g. when pressing the Message Waiting button).
##  Note: This parameter is supported
##  J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later.
##  PSTN_VM_NUM shall be used instead of MSGNUM in cases of IP Office
environment, 3PCC SIP environment or when there is failover from Aura environment
to a non-Aura server.
###################################################################################
##################################

set SUBSCRIBE_LIST_NON_AVAYA “message-summary”
##SET PSTN_VM_NUM "*97"

###################################################################################
##################################
## ENABLE_WEBSERVER specifies whether the HTTP/S WEB Server is enabled or
disabled.
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later; If the phone boots up in 3PCC environment
and ENABLE_WEBSERVER is not explicitly set 0, it will be internally set to 1.
##        This is to enable web server by default in 3PCC environments.
## SET ENABLE_WEBSERVER 0
###################################################################################
##################################

SET ENABLE_WEBSERVER 1

###################################################################################
##################################
##
## DIALPLAN specifies the dial plan used in the telephone.
##  It accelerates dialing by eliminating the need to wait for
##  the INTER_DIGIT_TIMEOUT timer to expire.
##  The value can contain 0 to 1023 characters; the default value is null ("").
##  See the telephone Administrator's Guide for format and setting alternatives.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP
R2.0.0.0 and later
#
## SET DIALPLAN [23]xxxx|91xxxxxxxxxx|9[2-9]xxxxxxxxx
###################################################################################
##################################
SET DIALPLAN [23]xxxx|91xxxxxxxxxx|9[2-9]xxxxxxxxx
SET 3PCC_SERVER_MODE 0

SET DNSSRVR 8.8.8.8
SET SSH_ALLOWED 1
SET AUTHCTRLSTAT 1
SET TLSSRVRID 0

###################################################################################
##################################
```

```
## GET $MACADDR will request for the "MACADDR" file from the HTTP/HTTPS Server
where "$MACADDR" which will be replaced by the telephone's MAC address.
##  Note: This parameter is supported by J129 SIP  R1.1.0.0, J169/J179 R1.5.0,
J100 SIP R2.0.0.0 and later
##################################################################################
##################################

GET $MACADDR.txt
```

Device-specific File: <macaddress>.txt

NOTE: This is an example file and it should be used for reference only.

```
SET FORCE_SIP_USERNAME 1001
SET FORCE_SIP_EXTENSION 1001
SET FORCE_SIP_PASSWORD 123456
SET CONFERENCE_FACTORY_URI "nway@10.133.46.166"
```

References

[1] Avaya, Inc. 2018. *Installing and Administering Avaya J100 IP Phone in Third-Party Call Control Setup*, *Release 4.0.1.0* Available from Avaya at support.avaya.com.