



Avaya J100 Series IP Phones

Partner configuration guide for 3CX



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Introduction to J100 Series IP Phones

Avaya J100 Series IP Phones provide a range of applications and features for unified communications. The phones leverage the enterprise IP network and eliminate the need of a separate voice network. The phones offer superior audio quality with the amplified handsets and customization with low power requirements in a Session Initiation Protocol (SIP) environment.

The phones can be configured

Supported models



Deployment

Avaya J100 IP Phones with Device Enrolment Service (DES)

To ease deployment and to be able to plug in the phone right out of the box and have the phone automatically configure and login it is recommended to use the Device Enrolment Service (DES). DES allows the phone to be automatically pointed to the appropriate Provisioning Server (either custom provisioning server (below) or 3CX server). To use this service, you must have an account on DES. Please see *“Installing and Administering Avaya J100 series IP Phone in third-party call control setup”* for more information on the DES. Alternatively, the phone can be pointed to the Provisioning Server via DHCP.

Avaya J100 IP Phones with a custom Provisioning Server

You can configure a custom Provisioning Server and use it to provide the configuration files to the phones. The provisioning server hosts the `J100Supgrade.txt` file, the `J100settings.txt` file as well as other configuration files (e.g. language, images, certificates) needed to get the phone configured and connected to the network.

Avaya J100 IP Phones with 3CX server

With 3CX server, you can automatically create the appropriate device (`J100settings.txt`) and user (`<MACaddress>.txt`) specific configuration files from the management system on the 3CX Provisioning Server. In this configuration, the phone does not require a custom Provisioning Server and connects to the 3CX Provisioning Server instead. For more information specifically related to the 3CX Provisioning service please see the *3CX support portal* <https://www.3cx.com/support/>

Provisioning server

The Provisioning Server is an HTTP or an HTTPS server that hosts the phone firmware and configuration files.

Files on the Provisioning Server

File name	Content
J100Supgrade.txt	Contains pointers to the firmware and upgrade files
J100setting.txt	Contains the configurable parameters that apply to all devices in a given deployment
<MACaddress>.txt	Contains configurable parameters that apply to an individual device in a deployment (typically the username, extension and password)
Resource files	e.g. languages, background and screen saver images, ringtones, trust certificate files,

Phone configuration

You can configure Avaya J100 Series IP Phones to work in the 3CX environment in the following ways:

- **Centralized configuration using file server**
- **Configuration through the web interface**

Please see *“Installing and Administering Avaya J100 series IP Phone in third-party call control setup”* for more information on the Configuration through the web interface.

Centralized configuration

You can bulk configure Avaya J100 Series IP Phones by using the 46xxsettings file. The phone can access the settings file via HTTP and HTTPS.

The following configuration files must be available on the file server configured for the devices.

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 J129 IP Phone to work with. Subscriber-specific settings are described

in the next section. For parameter description, see the *Installing and Administering Avaya J129 IP Phone in third-party call control setup* for J129.

1.1.1 Configure Network Settings

Step	Command
Step 1	SET ENABLE_UDP_TRANSPORT 1
Step 2	SET DNSSRV "8.8.8.8"
Step 3	SET DOMAIN ""
Step 4	SET SNTPSRVR pool.ntp.org
Step 5	SET SNTP_SYNC_INTERVAL 144000

1.1.2 Configure SIP Interface Settings

Step	Command
Step 1	SET SIPDOMAIN "3cx.hcm.com"
Step 2	SET SIP_CONTROLLER_LIST "3cx.hcm.com:5060;transport=tcp" (or udp, tls)
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 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

1.1.3 Configure Service Settings

Step	Command
Step 1	SET DIALPLAN [23]xxxx 91xxxxxxxxxxx 9[2-9]xxxxxxxxxx 7xxx
Step 2	SET NO_DIGITS_TIMEOUT 20
Step 3	SET INTER_DIGIT_TIMEOUT 5
Step 4	SET CALLFWDSTAT 7
Step 5	SET CALLFWDDELAY 1
Step 6	SET ENABLE_DND 1
Step 7	SET ENABLE_DND_PRIORITY_OVER_CFU_CFB 0
Step 8	SET ENABLE_AUTO_ANSWER_SUPPORT 1
Step 9	SET AUTO_ANSWER_MUTE_ENABLE 1
Step 10	SET HOLD_REMINDER_TIMER 0
Step 11	SET CONFERENCE_FACTORY_URI "conference@as.iop2..net"
Step 12	SET SIPCONFERENCECONTINUE 0
Step 13	SET PSTN_VM_NUM "*62"
Step 14	SET SUBSCRIBE_LIST_NON_AVAYA "message-summary"
Step 15	SET RINGTONESTYLE 0

1.1.4 Configure J100 Settings file

Step	Command
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
Step 6	SET SIPDOMAIN "3cx.hcm.com"
Step 7	SET TRUSTCERTS 3cx.pem
Step 8	SET SIP_CONTROLLER_LIST "3cx.hcm.com:5060;transport=tcp" (or udp, tls)

1.2 SIP Feature Configuration

This section provides configuration instructions for SIP features supported by the phone.

1.2.1 Registration and P2Pcall with 3CX

TCP configuration

Configure J100 Settings file

Step	Command
Step 1	SET SIP_CONTROLLER_LIST "3cx.hcm.com:5060;transport=tcp"

In 3Cx server

Step 1: Go to Extensions > "Edit extension xxxx" > Phone Provisioning > SIP Transport: TCP

The screenshot shows the 'Network' section of the 3CX web interface. Under 'Network interface for registration and provisioning', the IP address '10.16.29.11' is entered. Below this, the 'SIP Transport' section shows a list of options: TCP, UDP, TCP (highlighted in blue), and TLS.

TLS configuration

Configure J100 Settings file

Step	Command
Step 1	SET SIP_CONTROLLER_LIST "3cx.hcm.com:5061;transport=tls"
Step 2	SET TRUSTCERTS 3cx.pem

In 3Cx server

Go to Extensions > "Edit extension xxxx" > Phone Provisioning > SIP Transport: TLS

UDP configuration

Configure J100 Settings file

Step	Command
Step 1	J100 settings file: SET SIP_CONTROLLER_LIST "3cx.hcm.com:5060;transport=udp"

In 3CX server

Go to Extensions > "Edit extension xxxx" > Phone Provisioning > SIP Transport: UDP

1.2.2 Transfer

Attended transfer and blind transfer is supported in J100 Series IP Phones.

1.2.3 Call pickup

Go to Settings > PBX > General Options > Limit Call Pickup to calls received in their extension group:

You can see that limit Call Pickup to calls received in their extension group is not selected -> calls can only be answered among extensions that are part of the same extension group. The extension group has the options "Can see group members", "Can see group calls" and "Perform Operations (divert, transfer, pickup)" these are selected under the extension's "Rights" tabConference.

General Options

- ☒ Allow forwarding to external numbers in Ring Groups and Queues
- ☒ Play Busy prompt when extension is busy
- ☐ Limit Call Pickup to calls received in their extension group

Select the Limit Call Pickup to calls received in their extension group: you will be able to pick up calls ringing on other extensions regardless of extension group membership and what is checked under the extension's rights tab

General Options

- ☒ Allow forwarding to external numbers in Ring Groups and Queues
- ☒ Play Busy prompt when extension is busy
- ☒ Limit Call Pickup to calls received in their extension group

Groups > Add => create new group

Groups

Extension Groups

+ Add

Edit

Delete

Group name

DEFAULT

New group

OKCancel

GeneralGroup Rights

Group Name

New group

Groups > "edit group xxx" > Members > Add => add extensions to group

Group Name

Hoang_Pickup_1

Members

+ Add

Extension

First Name

70001

Hoang 70001

Extensions > "Edit extension xxxx" > Rights > check the box: Perform operations (divert, transfer, pickup) => allow permission to pick up for users

Rights

Group Membership

DEFAULT

Role

User

☒ Can see group members

☒ Can see group calls

☒ Show presence to group members

☒ Show calls to group members

☒ Perform operations (divert, transfer, pickup)

Settings > Parameters > "PICKUP": *20* => set pickup code

Parameter Settings

Custom Parameters

This page allows editing of advanced options. Only use this after having been recommended to do so by 3CX Technical Support. ADVANCED USERS ONLY!

+ Add

pickup dial code

Name	Description	Value
PICKUP	Pickup dial code	*20*

1.2.4 Call Forward

Configure J100 Settings file

Step	Command
Step 1	SET CALLFWDSTAT 7 => display call forward, call forward-Busy, call forward-No Answers

In 3CX server

Extensions > "Edit extension xxxx" > Forwarding Rules > Statuses:

Available/Away/DND/Lunch/Business Trip => choose status you want to configure

Extensions > "Edit extension xxxx" > Forwarding Rules > Statuses: Available > If I do not answer calls within: 10 Seconds and If I am busy or my phone is unregistered, forward calls to:

- Forward to Voicemail
- Forward to extensions
- End call

General Voicemail **Forwarding Rules** Phone Provisioning BLF

Statuses

If in this status, forward calls as follows

Available

Internal Calls

If I do not answer calls within: 20 seconds. Forward internal calls to:

Forward to Voicemail

Forward to Voicemail

Forward to extension's Voicemail

Forward to Mobile

Forward to Extension

Forward to number

End Call

Extensions > "Edit extension xxxx" > Web Authentication > Enable Web client and Password - Username is extension number: "132456" => enable and set password for web client

Web Authentication

i You can view the presence of your colleagues, c

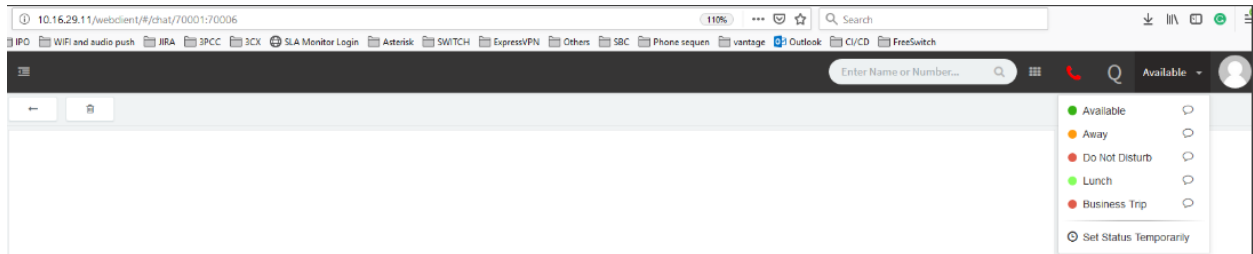


Enable Web client

Password - Username is extension number

••••••••••••••••••••

On web browser: open web client of 3CX <http://10.16.29.11/webclient/> and log in extension => choose status for extension



1.2.5 Conference

Configure J100 Settings files to default values.

Do not set "CONFERENCE_FACTORY_URI" on settings file

Meet-me conference

Settings > Conferencing > Audio > Conference Extension: 50000 and Require Conference PIN: 11111 => configure meet-me number and PIN

Conferencing Settings

OK
Cancel

Video
Audio

Audio Conferencing

Conference Extension

50000

1.2.6 MWI

Configure J100 Settings file

Step	Command
Step 1	SET PSTN_VM_NUM 98
Step 2	SET SUBSCRIBE_LIST_NON_AVAYA message-summary

In 3CX server

Extensions > "Edit extension xxxx" > Voicemail > Enable Voicemail and PIN Number: 123456

General
Voicemail
Forwarding Rules

☒
Enable Voicemail

Voicemail Language

Standard English Prompts Set

PIN Number

1431

Extensions > "Edit extension xxxx" > Forwarding Rules > Statuses: Available > If a call is unanswered for 10 Seconds or the user's phone is unregistered, then calls can be forwarded to voicemail:

Forward to Voicemail

The screenshot shows the 'Forwarding Rules' configuration page. At the top, there are five tabs: 'General', 'Voicemail', 'Forwarding Rules' (which is selected), 'Phone Provisioning', and 'BLF'. Below the tabs is a section titled 'Statuses'. Under this section, it says 'If in this status, forward calls as follows'. There is a dropdown menu showing 'Available'. Below that is a section titled 'Internal Calls'. It says 'If I do not answer calls within: 20 seconds. Forward internal calls to:'. There is a dropdown menu with the following options: 'Forward to Voicemail' (which is highlighted in blue), 'Forward to extension's Voicemail', 'Forward to Mobile', 'Forward to Extension', 'Forward to number', and 'End Call'.

Settings > Voicemail Settings > Voicemail Menu Extension Number: 99999

The screenshot shows the 'Voicemail Settings' dialog box. At the top, there is a title bar with the text 'Voicemail Settings' and two buttons: 'OK' and 'Cancel'. Below the title bar is a section titled 'Voicemail Menu'. Under this section, there is a label 'Voicemail Menu Extension Number' and a text input field containing the number '99999'.

1.2.7 Emergency call

Configure J100 Settings file

Step	Command
Step 1	SET ENABLE_SHOW_EMERG_SK 2
Step 2	SET ENABLE_SHOW_EMERG_SK_UNREG 2
Step 3	SET PHNEMERGNUM 911

NOTE:

- Phone can't end the emergency call on 3CX system
- Phone can't make an emergency call by pressing "Emerg" Softkey when logging out

1.2.8 DND support

Configure J100 Settings file

Step	Command
Step 1	SET ENABLE_DND 1

In 3CX server

Extensions > "Edit extension xxxx" > Forwarding Rules > Statuses: DND > If I am away forward internal calls to:

- Forward to Voicemail
- Forward to extensions
- End call

General
Voicemail
Forwarding Rules

Statuses

If in this status, forward calls as follows

Do Not Disturb (DND)

Internal Calls

If I am away forward internal calls to:

Forward to Voicemail

Extensions > "Edit extension xxxx" > Web Authentication > Enable Web client and Password - Username is extension number: "132456" => enable and set password for web client

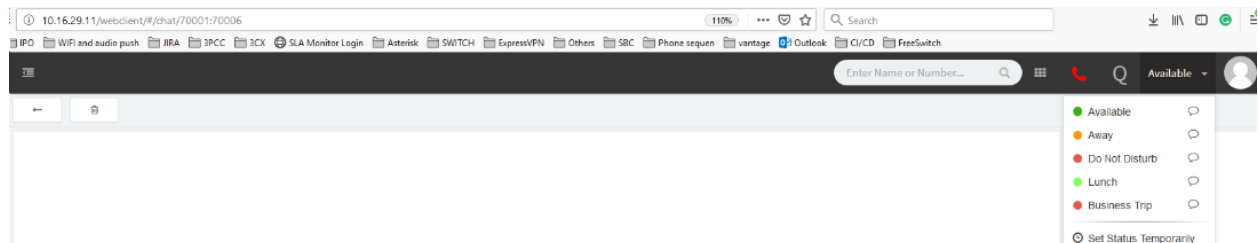
Web Authentication

You can view the presence of your colleagues, d

☒ Enable Web client

Password - Username is extension number

On web browser: open web client of 3CX <http://10.16.29.11/webclient/> and log in extension => choose status "DND" for extension

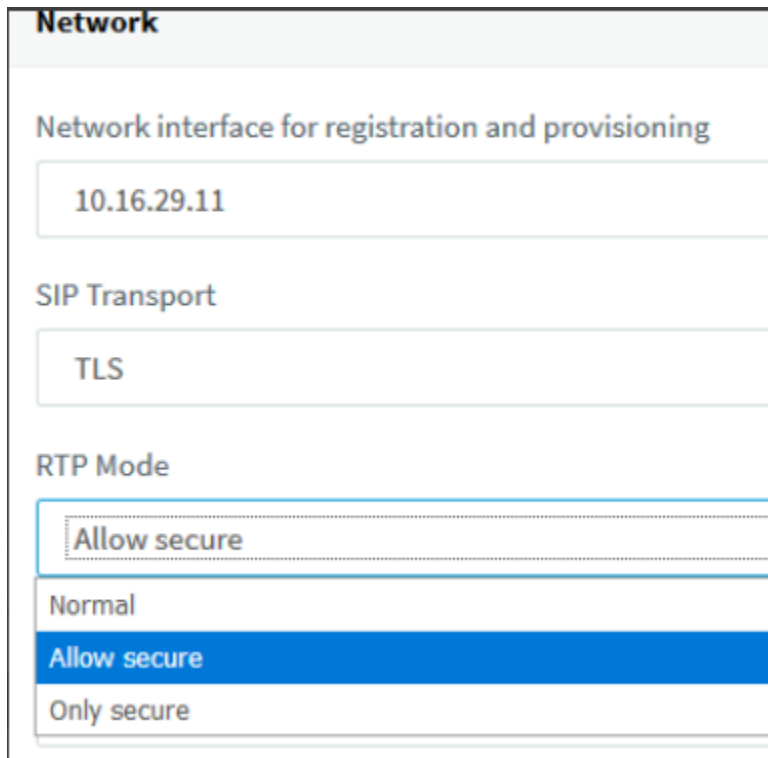


1.2.9 SRTP call support

Extensions > "Edit extension xxxx" > Phone Provisioning > SIP Transport: TLS

Extensions > "Edit extension xxxx" > Phone Provisioning > RTP Mode:

- **Allow Secure** => This will allow Secure RTP and Non-Secure RTP
- **Only Secure** => This will ONLY allow Secure RTP Connections



The screenshot shows a configuration page titled "Network". Under the heading "Network interface for registration and provisioning", the IP address "10.16.29.11" is entered. Below this, under "SIP Transport", the value "TLS" is selected. Under "RTP Mode", a dropdown menu is open, showing four options: "Normal", "Allow secure" (which is highlighted in blue), and "Only secure".

NOTE:

3CX doesn't support crypto 3-8, 9-11 (only support crypto 1 and 2)

- Configuration:
 - No RTP encryption is needed - Set "Normal" on the PBX and disable it on the phone
 - RTP encryption is needed - Set "Only Secure" on the PBX and Mandatory (Compulsory) on the phone
 - The option "Allow Secure" that we have on the PBX, is used only for the 3CX clients.
- 3CX does NOT support negotiation for SRTP/RTP with hard phones. For a J100 phone, users must pick either forced RTP or forced SRTP (set media encryption 9 for RTP, media encryption 1 or media encryption 2 for SRTP)

1.2.10 Call park and unpark

Settings > Parameters > PARK: *0, UNPARK: *1 => set dial codes for park and unpark

Parameter Settings

Custom Parameters

This page allows editing of advanced options. Only use this after having been recommended to do so by 3CX Technical Support. ADVANCED USERS ONLY!

[+ Add](#)

park

Name	Description	Value
PARK	Park dial code	*0
UNPARK	Unpark dial code	*1

Settings > Call Parking > Automatically unpark forgotten calls

- Transfer to the user that originally parked the call
- Extension
- Send call to Voice mail of extension

Call Parking Settings

[OK](#) [Cancel](#)

☒ **Automatically unpark forgotten calls**

If a call is parked for more than: seconds, then:

Extension

Transfer to the user that originally parked the call

Extension

Send call to Voicemail of extension

Forward to Outside Number

NOTE:

Phone doesn't hear music on hold after retrieving the parked call if the waiting phone is holding the call. Please refer to the below scenario for more details:

1. Phone A calls phone B. Phone B answers the call => A and B are in an active call
2. Phone A parks the call to orbit 1 by making a blind transfer to *01 => Phone A parks the call successfully. Phone B hears parking music
3. Phone B holds the call => Phone B holds the call successfully
4. Phone C retrieves the parked call by dialing *11=> Phone C retrieves the parked call successfully but doesn't hear music on hold

Users Impact:

1. During the waiting, the far-end user retrieves the parked call, the waiting user may have an important incoming call. he will answer the second call and the parked call will be held successfully
2. After retrieving the parked call, the far-end user will think that there is a trouble with speech path because he doesn't know the waiting user is holding the call. They will end the call before the waiting user resumes the call

1.2.11 Automatic redial/callback

Settings > PBX > Transfer Back on Busy > enter dial code and timeout

Transfer Back on Busy
Automatically transfers call back to the person that transferred the call initially if the destination is busy (works blind transfers only)
DialCode (Example *3*) to use for these blind transfers
<input type="text" value="*3*"/>
Timeout that caller must wait for answer before being transferred back
<input type="text" value="2"/>

1.2.12 Anonymous calling support

Settings > Blacklisted Numbers > Add > Incoming caller ID to be blocked: *anonymous => reject anonymous call

The image shows a web interface for managing a blacklist. In the background, there is a table titled 'Blacklisted Numbers' with a search bar and buttons for '+ Add', 'Edit', and 'Delete'. Below the table is a section for 'Blacklisted Caller ID' with the entry '*anonymous'. A modal form titled 'Blacklist' is open in the foreground. It contains two main fields: 'Incoming caller ID to be blocked' which has the value '*anonymous', and a 'Description' field which is currently empty.

Blacklist	
Incoming caller ID to be blocked	*anonymous
Description	

1.2.13 DTMF support

Extensions > "Edit extension xxx" > Phone Provisioning > DTMF Mode:

- RFC2833: out-of-band (using RFC 2833 procedures)
- In-Band
- SIP INFO (is not supported by phone J100)

Network

Network interface for registration and provisioning

10.16.29.11

SIP Transport

TLS

RTP Mode

Normal

DTMF Mode

RFC2833

RFC2833

In-Band

SIP INFO

NOTE:

- For RTP call: DTMF In-Band doesn't work with codecs G722, OPUS (works with G711A, G711U, G729)
- For SRTP call: DTMF In-Band doesn't work with all codecs

1.2.14 Busy Lamp Field Configuration

Avaya J129 currently does not support this feature. Please see *"Installing and Administering Avaya J100 series IP Phone in third-party call control setup"* for more information on the Busy Lamp Field Configuration.

1.2.15 Feature Key Synchronization Configuration

Avaya J129 currently does not support this feature.

1.2.16 Emergency Call Configuration

Avaya J129 currently does not support this feature.

1.2.17 User Service Configuration

Avaya J129 currently does not support this feature.

1.2.18 Directory Configuration

Avaya J129 currently does not support this feature.

1.2.19 Call Logs Configuration

Avaya J129 currently does not support this feature.

1.2.20 Visual Voice Mail Configuration

Avaya J129 currently does not support this feature.