

PART 2

Unified Communications – VoIP Assessment



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Unified Communications – VoIP Assessment Part 2

About this assessment:

These assessment tasks provide an opportunity to demonstrate the competencies covered in the VoIP - Unified Communications subject.

- You are allowed to refer to your text books, notes and the Internet during the Assessment.
- The documentation and research work must be entirely your own.
- By commencing this assessment, you confirm that you have read and agree to abide by the ACIT Academic Honesty Policy

Successful completion of this assessment contributes towards attaining competency in the following:

ICTTEN512 Design and implement an enterprise voice over internet protocol and a unified communications network

ICTNWK610 Design and build integrated VoIP networks

ICTPMG611 Prepare a detailed design brief

ICTTEN611 Produce an ICT network architecture design

ICTNWK529 Install and manage complex ICT networks

There are 3 parts to this assessment

1. Design and planning
2. Deployment
3. Question and answer

These assessment tasks provide an opportunity for you to demonstrate the competencies required to design, plan and deploy unified communications solutions.

Part 2 - Deployment

This part requires you to deploy an IP-private branch exchange (PBX) solution. You shall work in pairs for purpose of testing your solution.

You are provided with:

1. An Asterisk/FreePBX distro ISO file
2. A PC with Internet access
3. Access to a VMWare VSphere infrastructure host in which you can create new VM's.

4. Asterisk and FreePBX installation and configuration documentation
5. IP addressing scheme, extension numbering scheme, and DID
6. Head office trunk name and password

Note: you are required to take screen shots of each stage of your implementation. These will form part of your submission.

Scenario

OTS has implemented an Asterisk PBX at their head office. A SIP trunk has been provisioned to this site and is now fully operational. OTS now wishes to deploy IP telephony to their regional offices. It is much more economical to have one large SIP trunk than many small SIP trunks and therefore the entire organization will use the SIP trunk provisioned to the head office. The regional offices are connected to the head office via site to site VPN's. Some regional offices are connected to each other via site to site VPN's. OTS has decided that each regional office will have its own PBX. Calls to external numbers and other OTS offices that are not directly connected will be trunked via the head office.

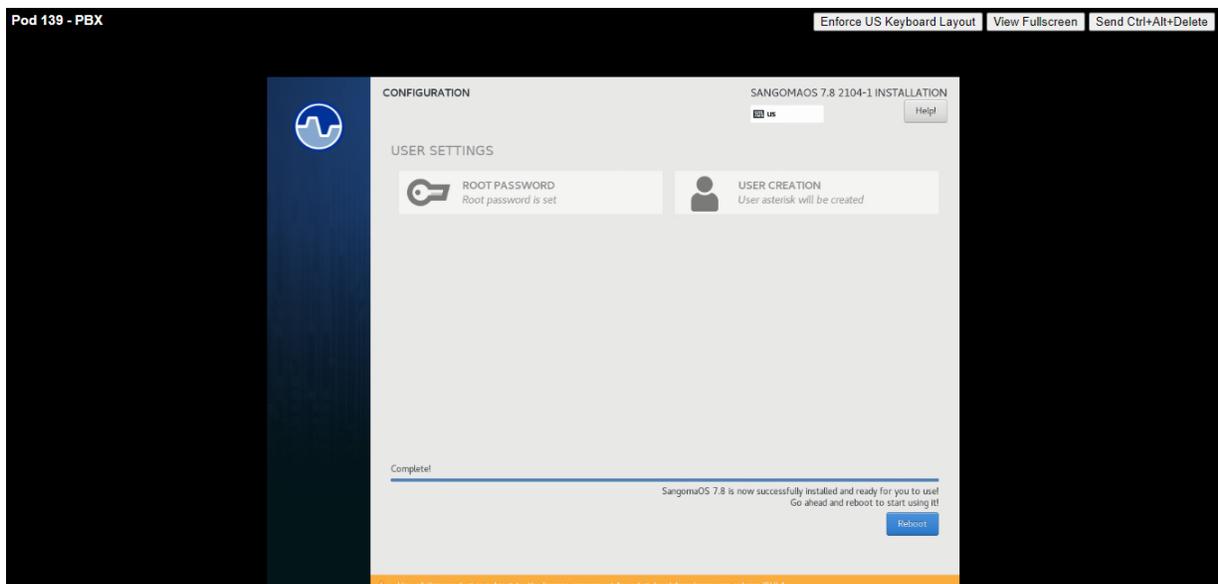
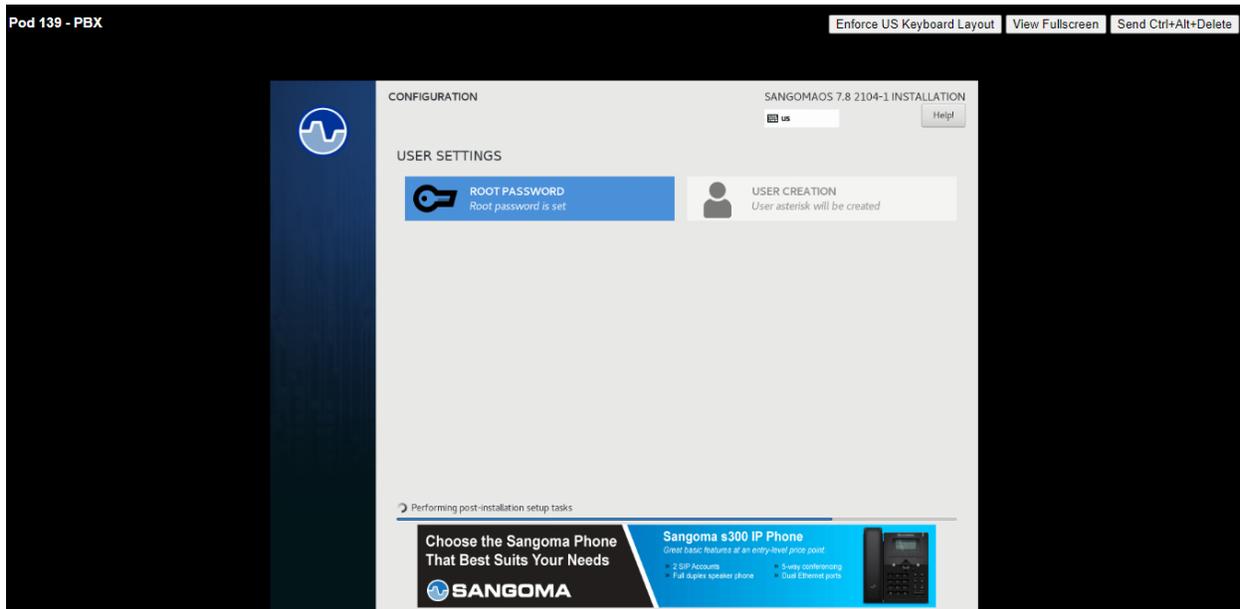
You have been given the task of designing and deploying the solution for a regional office. Your design and implementation plan must provide for the following:

A IP-private branch exchange (PBX) solution that will:

1. Provide extensions for the local office that can call each other
2. Provide an efficient low bandwidth trunk to the head office and one other office
3. Receive incoming calls from extensions in other offices
4. Receive incoming calls on the assigned DID
5. Make outgoing calls to other office extensions and external PSTN numbers
6. Provide call parking, IVR, call queuing, voice mail and a ring group

Task 1

Install Asterisk/FreePBX IP-private branch exchange (PBX) server and perform initial configuration. Use the IP addressing scheme provided. Record the IP address, username, and password of the server.



```
freepbx login: root
Password:
Last failed login: Wed Nov 24 12:55:29 AEST 2021 on tty1
There were 29 failed login attempts since the last successful login.
Last login: Wed Nov 3 13:02:22 on tty1
```



NOTICE! You have 7 notifications! Please log into the UI to see them!
Current Network Configuration

Interface	MAC Address	IP Addresses
eth0	00:50:56:87:55:62	10.10.41.139 fe80::250:56ff:fe87:5562

Please note most tasks should be handled through the GUI.
You can access the GUI by typing one of the above IPs in to your web browser.
For support please visit:
<http://www.freepbx.org/support-and-professional-services>

```
# FreePBX Sysadmin Generated network configuration.
# This file was generated at 2021-09-09T05:14:32+00:00
DEVICE=eth0
BOOTPROTO=static
ONBOOT='yes'
IPADDR=10.10.41.139
NETMASK=255.255.255.0
GATEWAY=10.10.41.1
~
~
```



x

System Admin

Network Settings

Wired Networks

Wireless Networks

Network Interface

eth0

IP Assignment

Static

DHCP

Unconfigured

Static IP

10.10.41.139

Netmask

24

Gateway

10.10.41.1

Start Automatically

Yes

No



Create Inte



System Admin

DNS

Normally, your first DNS server should be **127.0.0.1** . Add any additional servers after that.

DNS Server list

10.10.100.21
10.10.40.10

FreePBX Support | ISymphonyV3 Panel | UCP

Welcome to FreePBX Administration!

Initial Setup

Please provide the core settings that will be used to administer and update your system

Administrator User

Username

Password

Confirm Password

System Notifications Email

Notifications Email address

System Identification

System Identifier

System Updates

Automatic Module Updates

Automatic Module Security Updates

Send Security Emails For Unsigned Modules

Check for Updates every

Applications Connectivity Dashboard Reports Settings UCP

Activation

New Activation Existing Deployment

You should now enter a location name for this machine, This will be displayed on the FreePBX Dashboard, as well as in the Portal, to help identify this machine.

If you do not enter a name, one will be automatically generated.

You may be eligible for further offers after activation.
If there are any further offers, they will be displayed after you click 'Activate'.

Location Name:

Restart Activate

FreePBX is a registered trademark of Sangoma Technologies Inc. SANGOMA

Admin Applications Connectivity Dashboard Reports Settings UCP Apply Config

Sangoma Smart Firewall is now enabled!



To receive the full benefits of the Sangoma Smart Firewall, you should ensure that **no other firewall** is intercepting traffic to this machine. This is normally accomplished by configuring your internet connection to place this machine in the 'DMZ' of your gateway.
If you are unable to do this, it is unlikely that Responsive Firewall will work correctly, if at all.

Abort Continue

Admin Applications Connectivity Dashboard Reports Settings UCP

System Overview

Welcome to FreePBX
FreePBX 15.0.17.34 'VoIP Server Pod-139'
(You can change this name in Advanced Settings)

Summary Sysinfo updated 8 seconds ago

Asterisk	✓
MySQL	✓
Web Server	✓
Fail2Ban	✓
System Registration	✓
System Firewall	✓
Mail Queue	✗
Restapps Daemon	✗
UCP Daemon	✓
Xmpp Daemon	✓

Warnings Found
Please check for errors in the notification section

- Intrusion detection handling method
- Collecting Anonymous Browser Stats
- Default bind port for CHAN_PJSIP is: 5060, CHAN_SIP is: 5160

Show New

FreePBX - Let Freedom Ring Feed

- FreePBX 16 & Debian 11: An apt Combination
- User Control Panel Templates
- FreePBX 16 Beta is Here
- Introducing SmartOffice™ Access, A Convenient & Simple to Use Door Entry Solution for Small Businesses
- Springing toward Astricon 2021 - Call for Speakers
- How To Add a Remote MP3 Stream for Music on Hold

Sangoma Feed

- VI Communication Services' Voice for MS Teams
- Will 5G Fixed Wireless be viable to handle real-time UC?
- Save Money, Cut Your IT Costs & Reduce Complexity with Desktop-as-a-Service
- How SD-WAN Can Help Your Business
- CCaaS & Its Role In The Contact Center's Future
- Future of the Deskphone for Knowledge Workers

FreePBX Statistics

Asterisk Uptime CPU Memory Disk Network

Users Online Users Offline Trunks Online Trunks Offline Channels In Use

Live Network Usage

Interface eth0

Notepad

Pod 139	Daniel Ivan Cortez Olivares	10.10.41.139	0756200129
---------	-----------------------------	--------------	------------

```
username: daniel.cortez
password: Passw00rd
```

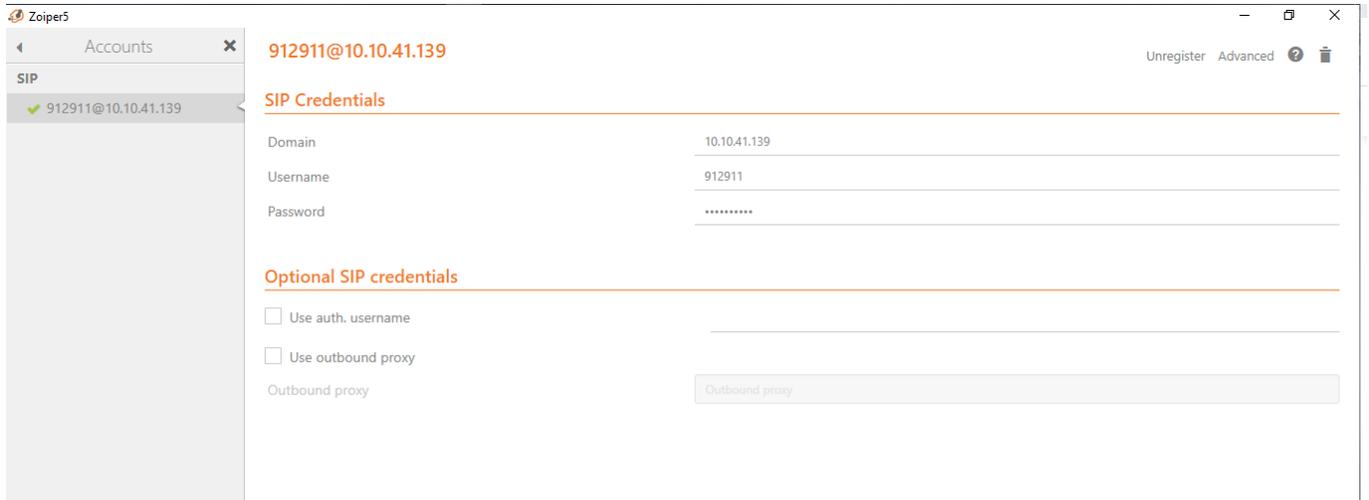
Task 2

Create 2 extensions in accordance with your extension numbering scheme and configure softphones or handsets for each of them. You may install and configure one softphone on the PC you are using and another on a mobile device that is connected to the campus network. If you are doing this assessment remotely you will need to create 2 client VM's and configure a softphone on each of them. Test the extensions to make sure they are working. Record the extension numbers.

The screenshot shows the 'Add PJSIP Extension' configuration page. The 'General' tab is selected. The 'Add Extension' section contains a note: 'This device uses PJSIP technology listening on Port 5060 (UDP)'. Below this are input fields for 'User Extension', 'Display Name', 'Outbound CID', 'Emergency CID', and 'Secret'. The 'Language' section has a 'Language Code' dropdown set to 'Default'. The 'User Manager Settings' section has a 'Select User Directory' dropdown set to 'PBX Internal Directory'. 'Submit' and 'Reset' buttons are at the bottom right.

The screenshot shows the 'All Extensions' management interface. It includes a search bar and buttons for '+ Add Extension', 'Quick Create Extension', and 'Delete'. The table below lists the configured extensions.

	Extension	Name	CW	DND	FM/FM	CF	CFB	CFU	Type	Actions
<input type="checkbox"/>	912911	daniel.cortez - laptop	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	
<input type="checkbox"/>	912912	daniel.cortez - mobile	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	



Zoiper5 - □ ×

✓ 912911@10.10.41.139 ⚙️

🔍 Find a contact .. ☰

Contacts Recent

All Online Favorites +

Click here to add a new contact 

Today

- Call to **Phone (912911)**, rejected. 3:44 PM ☑️
Busy Here (code: 486)
- Call from **Phone (912911)**, rejected. 3:44 PM ☑️

⋮

CHAT FEATURE IS UNAVAILABLE
 This **functionality** and many more useful features are available with Zoiper PRO

Learn more
Upgrade now

✓ ✉️ 🔊 📞 ⏪

Zoiper5 - □ ×

✓ 912911@10.10.41.139 ⚙️

🔍 Find a contact .. ☰

Calls

 daniel.cortez - mo...
 912911@10.10.41.139
📶 00:11

Contacts Recent

All Online Favorites +

Click here to add a new contact 

Mute
Speaker
Keypad
Statistics
Record P
Video P
Hold
Transfer P
Add call P



daniel.cortez - mobile
 912912
 912911@10.10.41.139
 DNID: 912911
00:11

📞

📞 ✉️ 🔊 📞 ⏪



Task 3

Create VoIP trunks to the head office PBX and one other PBX (this is your other team member's PBX). The trunks must consume as little bandwidth as possible. The head office PBX IP address is 10.10.41.2. The trunk credentials are:

Between your FreePBX and your team member's PBX:

- **Username: [yourname.surname@PBX]**
- **Password: P@ssw0rd**

Between your FreePBX and head office PBX:

- **Username: [your FreePBX IP address]**
- **Password: P@ssw0rd**

Provide answers to the following:

What type of trunk did you create and why did you choose that type?

Between my FreePBX and my team member's PBX I created a IAX2 TRUNK, 2 which allows to create trunks between Asterisk PBX's. The advantage of using IAX2 is that it combines signaling media in the same protocol and compresses multiple calls to save bandwidth.

List the settings you configured under 'Peer Details' and describe what each of those settings does.

Add Trunk

General | Dialed Number Manipulation Rules | **iax2 Settings**

Trunk Name

Hide CallerID Yes No

Outbound CallerID

CID Options Allow Any CID Block Foreign CIDs Remove CNAM Force Trunk CID

Maximum Channels

Asterisk Trunk Dial Options
 Override System

Continue if Busy Yes No

Disable Trunk Yes No

Monitor Trunk Failures
 Yes No

>

Add Trunk

General | **Dialed Number Manipulation Rules** | **iax2 Settings**

Outgoing | Incoming

Trunk Name

PEER Details

```
host=10.10.41.200
username=daniel.cortez@PBX
secret=P@ssw0rd
type=friend
qualify=yes
qualifyfreqok=25000
transfer=no
trunk=yes
```

>

Peers Details: This is where the tunnel between both Pbx is created.

Outgoing:

host: Is the IP address of the remote system that you are going to connect to. You can also put the domain name. If the other system will register to your system (using the Registration String field on the remote system), you should put the word "dynamic" here.

Username: Is the username that will be sent to the remote system when you attempt to place a call to authenticate the call. If the remote system requires authentication on incoming calls, the username= on the local system must match the name put in the "Trunk Name" in the PEER details on the remote system.

secret: Is the password that will be sent to the remote system when you attempt to place a call to authenticate the call. It is also the password that you will expect to receive when you receive a call, unless insecure=invite is used in the PEER details

type: "friend" means that you will both send calls to and receive calls from this server and that the PEER details will be used both for incoming and outgoing calls on this trunk.

qualify: "yes" means that your system will periodically send a request that the other system identify itself. If there is no answer within 2 seconds, your system will assume that the other system is down and stop sending calls to the system until the system responds. Instead of "yes," you can also put a number, in milliseconds. For example, qualify=3000 means that your system will wait 3 seconds for a response, instead of 2 seconds.

qualifyfreqok: The qualifyfreqok setting determines how often to ping the peer when it's in an OK state.

transfer: You can set transfer to yes, no, or mediaonly. If set to yes, Asterisk will transfer the call away from itself if it can, in order to make the packet path shorter between the two endpoints. (This obviously won't work if Asterisk needs to transcode or translate between protocols, or if network conditions don't allow the two endpoints to talk directly to each other.) If it is set to no, Asterisk will not try to transfer the call away from itself.

trunk (channel): IAX2 trunking enables Asterisk to send media (as mini-frames) from multiple channels using a single header. The reduction in overhead makes the IAX2 protocol more efficient when sending multiple streams to the same endpoint.



Add Trunk

General | **Dialed Number Manipulation Rules** | iax2 Settings

Outgoing | Incoming

USER Context ⓘ

jaime.tafoya@PBX

USER Details ⓘ

```
secret=P@ssw0rd  
type=user  
context=from-trunk
```

Register String ⓘ

» Submit Reset

General | **Dialed Number Manipulation Rules** | iax2 Settings

Trunk Name ⓘ admin@PBX

Hide CallerID ⓘ Yes No

Outbound CallerID ⓘ 0756200129

CID Options ⓘ **Allow Any CID** Block Foreign CIDs Remove CNAM Force Trunk CID

Maximum Channels ⓘ

Asterisk Trunk Dial Options ⓘ T
Override System

Continue if Busy ⓘ Yes No

Disable Trunk ⓘ Yes No

Monitor Trunk Failures ⓘ Yes No

» Submit Res

General **Dialed Number Manipulation Rules** **iax2 Settings**

Outgoing Incoming

Trunk Name ⓘ Inter-Admin Trunk

PEER Details ⓘ

```
host=10.10.41.2
username=10.10.41.139
secret=P@ssw0rd
type=friend
qualify=yes
qualifyfreqok=25000
transfer=no
trunk=yes
```

General **Dialed Number Manipulation Rules** **iax2 Settings**

Outgoing Incoming

Trunk Name ⓘ Inter-Admin Trunk

PEER Details ⓘ

```
host=10.10.41.2
username=10.10.41.139
secret=P@ssw0rd
type=friend
qualify=yes
qualifyfreqok=25000
transfer=no
trunk=yes
```

Task 4

Configure outbound call routing so that calls can be made to extensions in another PBX. (This is the PBX of the other student you are working with). Verify that you can make and receive calls to and from the extensions in the other PBX.

Record the dial pattern that you are matching for this route and the name of the trunk through which calls will be forwarded.

Outbound Routes

Add Route

Route Settings
Dial Patterns
Import/Export Patterns
Notifications
Additional Settings

Route Name ⓘ

Route CID ⓘ

Override Extension ⓘ

Yes
No

Route Password ⓘ

Route Type ⓘ

Emergency
Intra-Company

Music On Hold? ⓘ

Time Match Time Zone: ⓘ

Time Match Time Group ⓘ

Trunk Sequence for Matched Routes ⓘ

+

ⓘ
🗑️

+

ⓘ
🗑️

Outbound Routes

Add Route

Route Settings
Dial Patterns
Import/Export Patterns
Notifications
Additional Settings

Dial Patterns that will use this Route

Pattern Help +

[Dial patterns wizards](#)

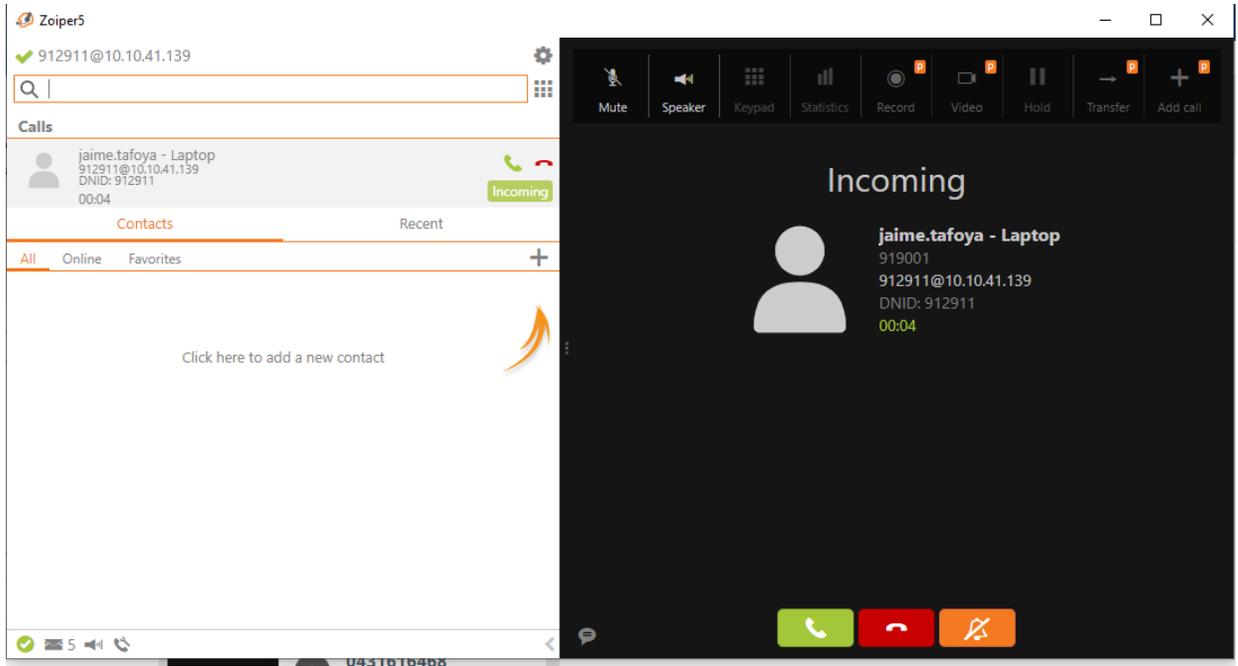
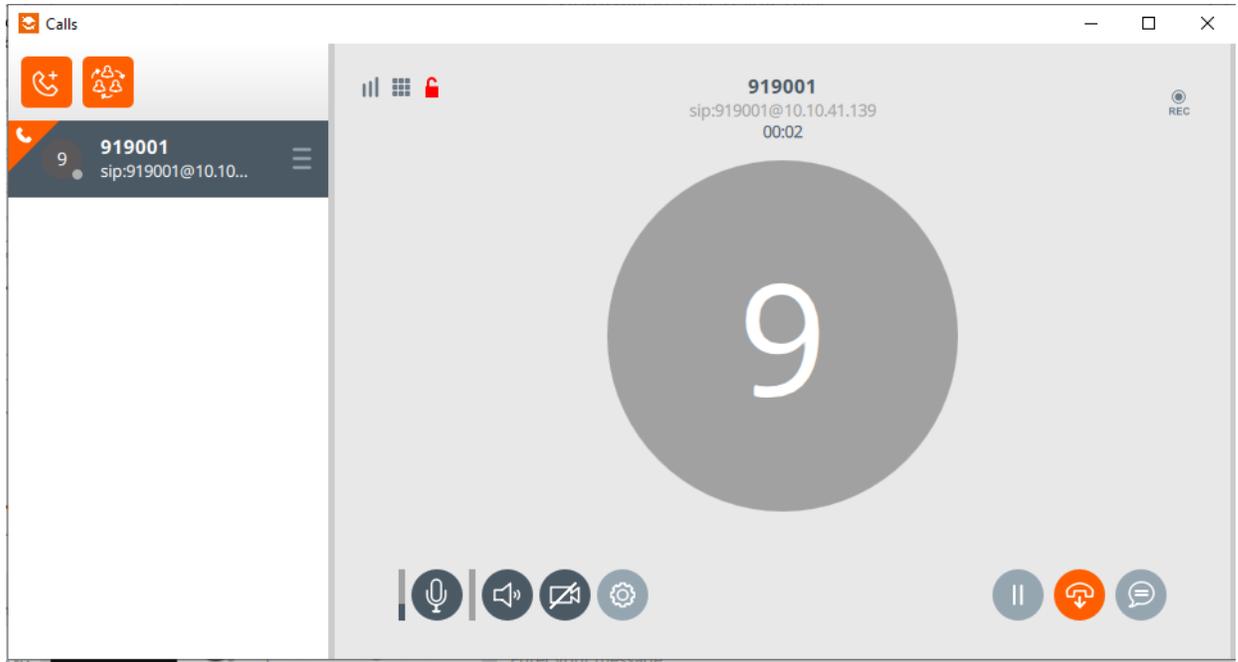
+

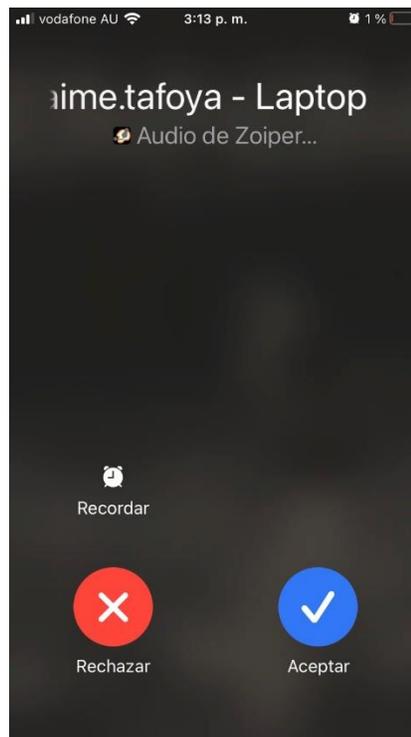
Outbound Routes

This page is used to manage your outbound routing.

[+ Add Outbound Route](#)

Name	Outbound CID	Attributes	Actions
+ daniel.cortez@Outbound	0756200129		
+ jaime.tafoya@Outbound			



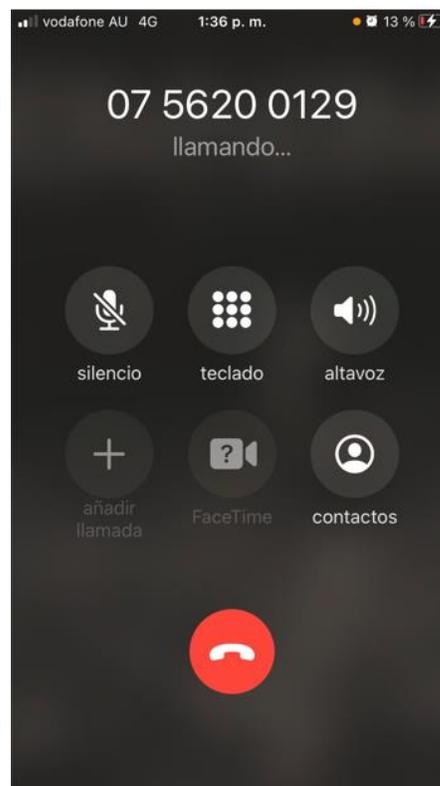


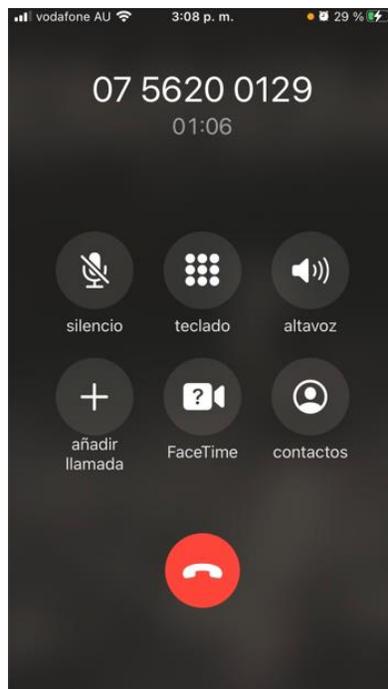
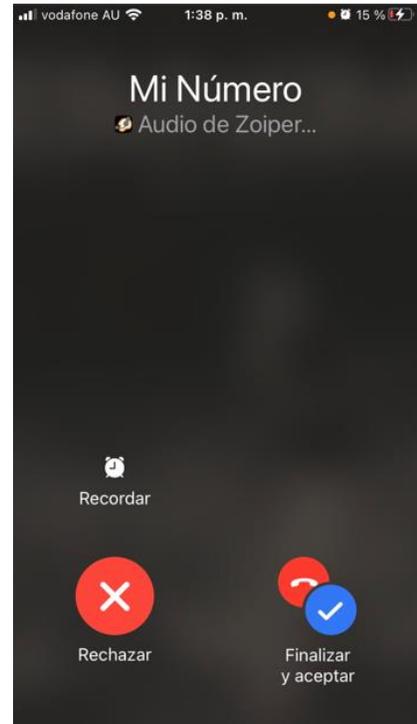
Task 5

Configure an inbound route so that calls can be received on your DID. Verify that you can receive calls with 2-way audio made from the PSTN. (you can do this by calling your DID from your mobile or any other phone on the PSTN). Record the test procedure and results.

Add Incoming Route

General	Advanced	Privacy	Fax	Other	
Description	daniel.cortez@inbound				
DID Number	0756200129				
CallerID Number	ANY				
CID Priority Route	<input type="radio"/> Yes <input checked="" type="radio"/> No				
Alert Info	None				
Ringer Volume Override	None				
CID name prefix					
Music On Hold	Default				
Set Destination	Ring Groups				
	912900 all extensions				
				<input type="button" value="Submit"/>	<input type="button" value="Res"/>





Zoiper5

✓ 912911@10.10.41.139

Find a contact ..

Calls



0431616468
912911@10.10.41.139
DNID: 912911
00:07

Contacts

All Online Favorites

Zoiper5

✓ 912911@10.10.41.139

Find a contact ..

Calls

0431616468
912911@10.10.41.139
00:44

Contacts Recent

All Online Favorites

Task 6

Configure an outbound route so that you can make calls to any external number. Record the dial pattern that you are matching and the name of the trunk through which calls will be forwarded. How did you test that you could make calls to the PSTN?

Outbound Routes

Add Route

Route Settings | Dial Patterns | Import/Export Patterns | Notifications | Additional Settings

Route Name

Route CID

Override Extension

Route Password

Route Type

Music On Hold?

Time Match Time Zone:

Time Match Time Group

Trunk Sequence for Matched Routes

-
-
-

Outbound Routes

This page is used to manage your outbound routing.

[+ Add Outbound Route](#)

Name	Outbound CID	Attributes	Actions
+ daniel.cortez@Outbound	0756200129		
+ jaime.tafoya@Outbound			
+ PSTN@Outbound	0756200129		



Dial Patterns that will use this Route

Pattern Help			
Dial patterns wizards			
prepend	prefix	0[2378]NXXXXXX	CallerID
prepend	prefix	000	CallerID
prepend	prefix	001X	CallerID
prepend	prefix	04XXXXXXX	CallerID
prepend	prefix	0967XXXX	CallerID
prepend	prefix	130XXXXXXX	CallerID
prepend	prefix	13XXXX	CallerID
prepend	prefix	180XXXXXXX	CallerID
07	prefix	55XXXXXX	CallerID
07	prefix	56XXXXXX	CallerID
prepend	prefix	match pattern	CallerID

Submit Dupli

Zoiper5

912911@10.10.41.139

Find a contact ..

Calls

0431616468
912911@10.10.41.139
00:07

Contacts Recent

All Calls Messages

Today

- 0431616468
0431616468 4:41 PM
- 0450762524
0450762524 4:39 PM
- daniel.cortez - mobile
912912 4:18 PM
- daniel.cortez - laptop
912911 3:38 PM

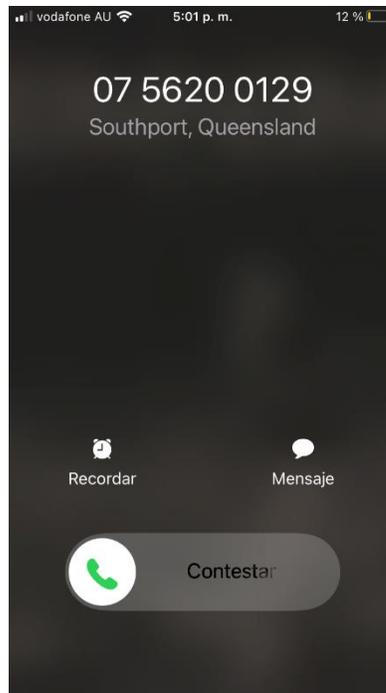
September 16, 2021

- gilberto.ferraz - Mobile
912702 3:47 PM

Mute Speaker Keypad Statistics Record Video Hold Transfer Add call

0431616468
0431616468
912911@10.10.41.139
00:07

4



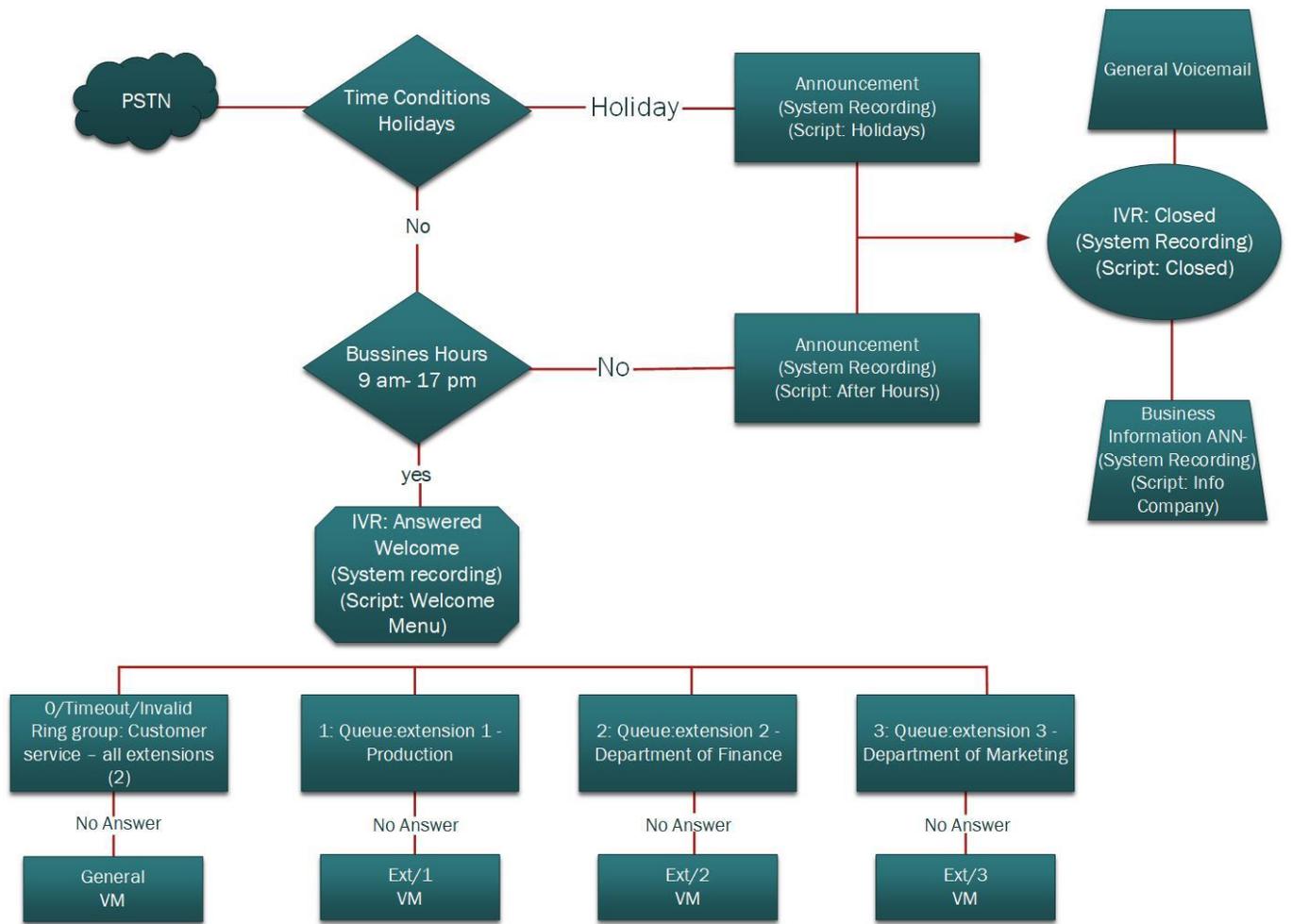
Task 7

Configure call parking, IVR, call queuing, voice mail, and a ring group.

How did you test that these functions are working?

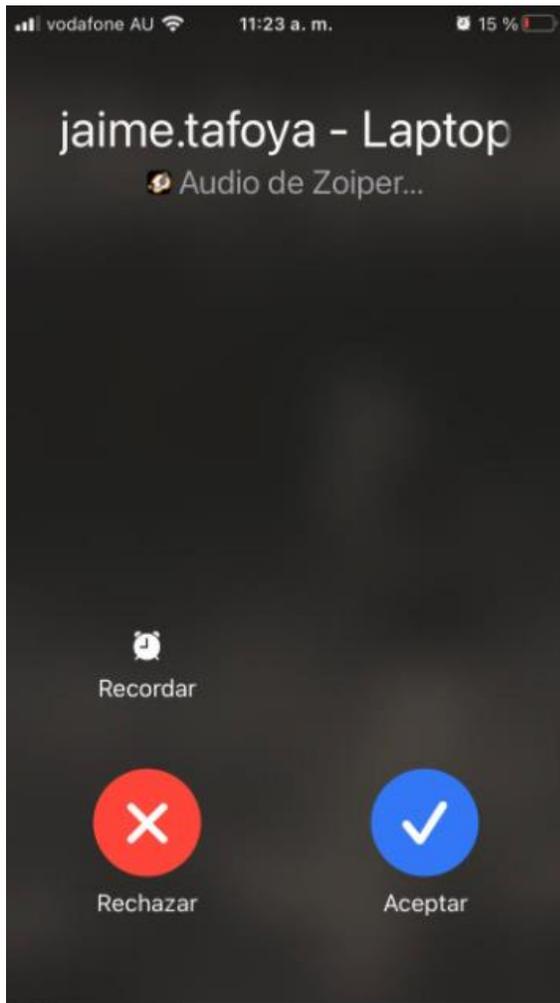
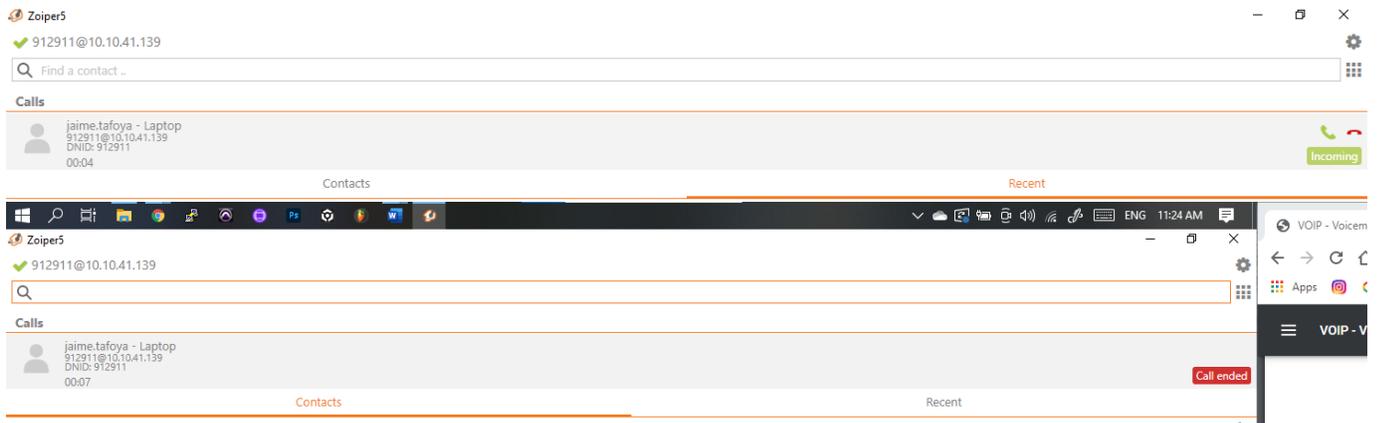
IVR





voicemail:





Zoiper5

912911@10.10.41.139

12345

Mute Speaker Keypad Statistics Record Video Hold Transfer Add call

Calls

jaime.tafoya - Mob...
912911@10.10.41.139
DNID: 912911
00:02 Incoming

Contacts Recent

All Online Favorites +

No contacts found that match your search
Would you like to add it?

Add

Incoming

jaime.tafoya - Mobile
919002
912911@10.10.41.139
DNID: 912911
00:02

You can use the collapse button

Got it

3

PBX Voicemail Notification ▷ Recibidos x



aste...@freepbx.sangoma.local

11:36 (hace 0 m)

para mí ▾

inglés ▾ > español ▾ [Traducir mensaje](#)

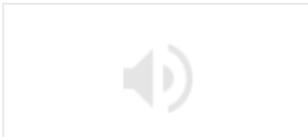
daniel.cortez - laptop,

There is a new voicemail in mailbox 912911:

From: "jaime.tafoya - Mobile" <919002>
Length: 0:09 seconds
Date: Thursday, November 04, 2021 at 11:35:05 AM

Dial *98 to access your voicemail by phone.

Visit <http://AMPWEBADDRESS/ucp> to check your voicemail with a web browser.



For announcements

(After Hours Message) – (ANN Script after hours) : “Thank you for calling the beats rap company, your call is important to us, we regret to inform you that our office is currently closed. Our business hours are from 9AM to 5PM, from Monday to Friday Australian Time” [GO TO IVR]

Pressed [2] - (ANN_BUSSINES_INFORMATION) –If you would like to know more about our company, you can visit us at beatsrap.com on your favorite browser, you can find related information about our prices and other products. Otherwise, you can leave as an email through beatsrap@gmail.com, and one of our representatives will get in touch with you within business hours. Thank you!” [HANG UP]

(ANN HOLIDAYS) “Thank you for calling the beatsrap, the best website to get your beat to make your amazing rap song, we are currently closed due to Holidays. We will return on the [10 JANUARY] at 9AM GOLD COAST TIME ” [GO TO IVR]

(In Queue Message) “Your call is important to us. All our representatives are still busy assisting other callers, we appreciate your patience. One of our agents will be with you shortly” [REPEAT EVERY 1 MINUTE(S)]

(ANN_goodbye) “Thank you for calling beatsrap, Goodbye”

(ANN_Invalid Key) “You pressed an Invalid key, here are the options again”

(IVR Timeout – Message) “Please stay on the line, one of our representatives will be with you shortly”

IVR For AFTER HOURS

[IVR CLOSED] “To leave a message in our voicemail, press 1: And A representative will contact you within a business day.” “Otherwise, Press 2 for more information about our business contacting methods”
[REPEAT 2 TIMES]

[IF NOT THEN GO TO NOT ANSWERED ANNOUNCEMENT]

IVR For Menu Options

(IVR_WELCOME_MENU_OK)“Thank you for calling Beatsrap. If you know your party’s extension you can dial it at any time, otherwise please listen carefully as our menu option has changed:

Press 1, If you want to contact the production department. [GO TO QUEUES]

Press 2 for department of finances [GO TO QUEUES]

To speak to one of our representatives of Marketing Department, press 3,
to speak to one of our representatives please stay on the line or press 0 to be redirected immediately. [GO TO RING GROUP]

If you would like to hear these options again press the Pound key #. [REPEAT IVR] [TIMEOUT/9/INVALID = RING GROUP]”

MUSIC ON HOLD

+ Add Recording

Search  

Display Name	Description	Supported Languages	Actions
ANN_bussines_information	SCRIPT: INFO COMPANY	English	 
ANN_GOOD BYE	ANN_GOOD BYE	English	 
ANN_HOLIDAYS	SCRIPT_HOLIDAYS	English	 
ANN_SCRIPT: AFTER HOURS	ANN_SCRIPT: AFTER HOURS	English	 
ANN_WELCOME	ANN_WELCOME	English	 
IN QUEUE MESSAGE	IN QUEUE MESSAGE	English	 
INVALID_KEY	INVALID_KEY	English	 
IVR_CLOSED	IVR_CLOSED	English	 
IVR_TIMEOUT_MESSAGE	IVR_TIMEOUT_MESSAGE	English	 
IVR_WELCOME_MENU_OK	IVR_WELCOME_MENU_OK	English	 

+ Add Category

Search

Category	Type	Actions
default	files	
MUSIC-ON-HOLD	files	 

Showing 1 to 2 of 2 rows

Extensions according to the company and ivr design

- All Extensions
- Custom Extensions
- DAHDI Extensions
- IAX2 Extensions
- SIP [chan_pjsip] Extensions
- SIP (Legacy) [chan_sip] Extensions
- Virtual Ext >

+ Add Extension  Quick Create Extension  Search   

<input type="checkbox"/>	Extension	Name	CW	DND	FM/FM	CF	CFB	CFU	Type	Actions
<input type="checkbox"/>	912901	Production	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	 
<input type="checkbox"/>	912911	Department of finances	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	 
<input type="checkbox"/>	912912	Department of Marketing	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	 
<input type="checkbox"/>	912999	General Voicemail	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	virtual	 

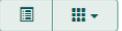
Showing 1 to 4 of 4 rows

QUEUES

Queues

[+ Add Queue](#)

Search



Queue	Description	Actions
912990	PRODUCTION queue	
912991	Department of finances queue	
912992	Department of Marketing queue	

Showing 1 to 3 of 3 rows

Queues Edit: 912990

Used as Destination by 1 Object (Click to Expand)

General Settings
Queue Agents
Timing & Agent Options
Capacity Options
Caller Announcements
Advanced Options
Reset Queue Stats

Queue Number 912990

Queue Name

Queue No Answer

Call Confirm

Call Confirm Announce

CID Name Prefix

Wait Time Prefix

Alert Info

Ringer Volume Override

Ringer Volume Override Mode

Restrict Dynamic Agents

Agent Restrictions

Ring Strategy

Autofill

Skip Busy Agents

Queue Weight

Music on Hold Class

Join Announcement

Call Recording

Mark calls answered elsewhere

Fail Over Destination

Fail Over Destination ? »

Queues Edit: 912990

Used as Destination by 1 Object (Click to Expand) ▼

General Settings Queue Agents Timing & Agent Options Capacity Options Caller Announcements Advanced Options Reset Queue Stats »

Static Agents ?

Dynamic Agents ?

Queues Edit: 912990

Used as Destination by 1 Object (Click to Expand) ▼

General Settings Queue Agents Timing & Agent Options Capacity Options Caller Announcements Advanced Options Reset Queue Stats »

Max Wait Time ?

Max Wait Time Mode ?

Agent Timeout ?

Agent Timeout Restart ?

Retry ?

Wrap-Up-Time ?

Member Delay ?

Agent Announcement ?

Report Hold Time ? »

Report Hold Time ?

If you want to report the caller's hold time to the member before they are connected to the caller, set this to yes.

Auto Pause ?

Auto Pause on Busy ?

Auto Pause on Unavailable ?

Auto Pause Delay ? »

Queues Edit: 912991

General Settings Queue Agents Timing & Agent Options Capacity Options Caller Announcements Advanced Options Reset Queue Stats

Queue Number 912991

Queue Name Department of finances queue

Queue No Answer Yes No

Call Confirm Yes No

Call Confirm Announce Default

CID Name Prefix Finances:

Wait Time Prefix Yes No

Alert Info None

Ringer Volume Override None

Ringer Volume Override Mode Force Yes Don't Care No Never

Restrict Dynamic Agents Yes No

Agent Restrictions Call as Dialed No Follow-Me or Call Forward Extensions Only

Ring Strategy rrmemory

Autofill Yes No

Skip Busy Agents No Yes Yes + (ringinuse=no) Queue calls only (ringinuse=no)

Queue Weight 0

Music on Hold Class MUSIC-ON-HOLD

MoH Only Agent Ringing Ring Only

Join Announcement None

Always When No Free Agents When No Ready Agents

Call Recording Force Yes Don't Care No Never

Mark calls answered elsewhere Yes No

Fail Over Destination Voicemail

912911 Department of finances (Unavailable Message)

Submit Reset Delete

Queues Edit: 912991

General Settings Queue Agents Timing & Agent Options Capacity Options Caller Announcements Advanced Options Reset Queue Stats

Static Agents 912911.0

Dynamic Agents

Agent Quick Select

Agent Quick Select

Queues Edit: 912991

General Settings	Queue Agents	Timing & Agent Options	Capacity Options	Caller Announcements	Advanced Options	Reset Queue Stats	>
Max Wait Time	5 minutes						>
Max Wait Time Mode	<input checked="" type="radio"/> Strict <input type="radio"/> Loose						
Agent Timeout	15 seconds						>
Agent Timeout Restart	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Retry	2 seconds						>
Wrap-Up-Time	15 seconds						>
Member Delay	0 seconds						>
Agent Announcement	None						>
Report Hold Time	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Auto Pause	<input type="radio"/> Yes in this queue only <input checked="" type="radio"/> Yes in all queues <input type="radio"/> No						
Auto Pause on Busy	<input type="radio"/> Yes <input checked="" type="radio"/> No						> <input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Delete"/>
Auto Pause on Busy	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Auto Pause on Unavailable	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Auto Pause Delay	0						> <input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Delete"/>

Queues Edit: 912992

General Settings	Queue Agents	Timing & Agent Options	Capacity Options	Caller Announcements	Advanced Options	Reset Queue Stats	>
Queue Number	912992						
Queue Name	Department of Marketing queue						
Queue No Answer	<input type="radio"/> Yes <input checked="" type="radio"/> No						
Call Confirm	<input type="radio"/> Yes <input checked="" type="radio"/> No						
Call Confirm Announce	Default						>
CID Name Prefix	marketing:						
Wait Time Prefix	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Alert Info	None						>
Ringer Volume Override	None						>
Ringer Volume Override Mode	<input type="radio"/> Force <input type="radio"/> Yes <input checked="" type="radio"/> Don't Care <input type="radio"/> No <input type="radio"/> Never						
Restrict Dynamic Agents	<input checked="" type="radio"/> Yes <input type="radio"/> No						
Agent Restrictions	<input type="radio"/> Call as Dialed <input checked="" type="radio"/> No Follow-Me or Call Forward <input type="radio"/> Extensions Only						> <input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Delete"/>

Ring Strategy [?](#)

Autofill [?](#)

Skip Busy Agents [?](#)

Queue Weight [?](#)

Music on Hold Class [?](#)

Join Announcement [?](#)

Call Recording [?](#)

Mark calls answered elsewhere [?](#)

Fail Over Destination [?](#)

Queues Edit: 912992

General Settings **Queue Agents** **Timing & Agent Options** **Capacity Options** **Caller Announcements** **Advanced Options** **Reset Queue Stats** [C](#) [>](#)

Static Agents [?](#)

Dynamic Agents [?](#)

Agent Quick Select

Agent Quick Select

Queues Edit: 912992

General Settings **Queue Agents** **Timing & Agent Options** **Capacity Options** **Caller Announcements** **Advanced Options** **Reset Queue Stats** [>](#)

Max Wait Time [?](#)

Max Wait Time Mode [?](#)

Agent Timeout [?](#)

Agent Timeout Restart [?](#)

Retry [?](#)

Wrap-Up-Time [?](#)

Member Delay [?](#)

Agent Announcement [?](#)

Report Hold Time [?](#)

Auto Pause [?](#)

Auto Pause on Busy [?](#)

Auto Pause on Unavailable Yes No

Auto Pause Delay »

IVR

IVR Name	IVR Description	Actions
CLOSED	CLOSED	<input type="button" value="edit"/> <input type="button" value="delete"/>
Welcome	Welcome	<input type="button" value="edit"/> <input type="button" value="delete"/>

Showing 1 to 2 of 2 rows

Edit IVR: Welcome

Used as Destination by 1 Object (Click to Expand) ▼

IVR General Options

IVR Name

IVR Description

IVR DTMF Options

Announcement ▼

Enable Direct Dial ▼

Force Strict Dial Timeout Yes No No - Legacy

Timeout

Alert Info ▼

Ringer Volume Override ▼

Invalid Retries

Invalid Retry Recording ▼

Append Announcement to Invalid Yes No

Return on Invalid Yes No

Invalid Recording »



Invalid Recording 🔗

Invalid Destination 🔗

Timeout Retries 🔗

Append Announcement on Timeout 🔗

Return on Timeout 🔗

Timeout Recording 🔗

Timeout Destination 🔗

Return to IVR after VM 🔗

— IVR Entries

Digits 🔗	Destination 🔗	Return 🔗	Delete
<input type="text" value="0"/>	<input type="text" value="Ring Groups"/> <input type="text" value="912900 all extensions"/>		
<input type="text" value="#"/>	<input type="text" value="IVR"/> <input type="text" value="Welcome"/>	<input checked="" type="button" value="Yes"/> <input type="button" value="No"/>	
<input type="text" value="*"/>	<input type="text" value="Feature Code Admin"/>		

»

	<input type="text" value="Dial Voicemail <*98>"/>		
<input type="text" value="1"/>	<input type="text" value="Queues"/> <input type="text" value="912990 PRODUCTION queue"/>		
<input type="text" value="2"/>	<input type="text" value="Queues"/> <input type="text" value="912991 Department of finances queue"/>		
<input type="text" value="3"/>	<input type="text" value="Queues"/> <input type="text" value="912992 Department of Marketing queue"/>		
<input type="text" value="digits pressed"/>	<input type="text" value="== choose one =="/>	<input checked="" type="button" value="Yes"/> <input type="button" value="No"/>	

+Add Another Entry

»



Edit IVR: CLOSED

Used as Destination by 1 Object (Click to Expand) ▼

— IVR General Options

IVR Name 🔗	CLOSED
IVR Description 🔗	CLOSED

— IVR DTMF Options

Announcement 🔗	IVR_CLOSED
Enable Direct Dial 🔗	Disabled
Force Strict Dial Timeout 🔗	<input type="radio"/> Yes <input type="radio"/> No <input checked="" type="radio"/> No - Legacy
Timeout 🔗	5
Alert Info 🔗	None
Ringer Volume Override 🔗	None
Invalid Retries 🔗	2
Invalid Retry Recording 🔗	INVALID_KEY
Append Announcement to Invalid 🔗	<input checked="" type="radio"/> Yes <input type="radio"/> No
Return on Invalid 🔗	<input checked="" type="radio"/> Yes <input type="radio"/> No
Invalid Recording 🔗	ANN_GOOD_BYE

» [Submit](#) [Duplicate](#) [Reset](#) [Delete](#)

Invalid Destination 🔗	Terminate Call Hangup
Timeout Retries 🔗	1
Timeout Retry Recording 🔗	None
Append Announcement on Timeout 🔗	<input checked="" type="radio"/> Yes <input type="radio"/> No
Return on Timeout 🔗	<input checked="" type="radio"/> Yes <input type="radio"/> No
Timeout Recording 🔗	ANN_GOOD_BYE
Timeout Destination 🔗	Terminate Call Hangup
Return to IVR after VM 🔗	<input type="radio"/> Yes <input checked="" type="radio"/> No

— IVR Entries

Digits 🔗	Destination 🔗	Return 🔗	Delete
1	Voicemail 912999 General Voicemail (Instructions Only)		
2	Announcements INFO_COMPANY		
digits pressed	== choose one ==	<input checked="" type="radio"/> Yes <input type="radio"/> No	

[+Add Another Entry](#)

» [Submit](#) [Duplicate](#) [Reset](#) [Delete](#)

Time Condition

Time Groups

[List Time Conditions](#) [+ Add Time Group](#) Server time: 16:29:44 AEST

Time Group	Actions
Business Hours	
Public Holiday	

Showing 1 to 2 of 2 rows

Time Groups

This time group is currently in use and cannot be deleted
Business Hours

Description

Time(s)

Time to Start	<input type="text" value="09"/>	<input type="text" value="00"/>
Time to finish	<input type="text" value="17"/>	<input type="text" value="00"/>
Week Day Start	<input type="text" value="Monday"/>	
Week Day finish	<input type="text" value="Friday"/>	
Month Day start	<input type="text" value="-"/>	
Month Day finish	<input type="text" value="-"/>	
Month start	<input type="text" value="-"/>	
Month finish	<input type="text" value="-"/>	

[+ Add Time](#) [» Submit](#) [Duplicate](#) [Reset](#) [Delete](#)

This time group is currently in use and cannot be deleted

Holidays

Description

Public Holiday

Time(s)

Time to Start

-

-

Time to finish

-

-

Week Day Start

-

-

Week Day finish

-

-

Month Day start

26

-

Month Day finish

26

-

Month start

January

-

Month finish

January

-

Time to Start

-

-

Time to finish

-

-

Week Day Start

-

-

Submit Duplicate Reset Delete

Time Conditions

List Time Groups + Add Time Condition Server time: 16:30:30 AEST Search

Time Condition	Override State	Linked Item	Actions
Business Hours	No Override	Time Group	
Holidays	No Override	Time Group	

Showing 1 to 2 of 2 rows

Time Conditions

Used as Destination by 1 Object (Click to Expand)

Edit Time Condition: Business Hours (*271)

Time Condition name

Business Hours

Override Code Pin

Invert BLF Hint

Yes No

Change Override

Unchanged

Current: Unknown State

Time Zone

Use System Timezone

Mode

Time Group Mode Calendar Mode

Time Group

Business Hours

Destination matches

Announcements

Welcome

Destination non-matches ⓘ

Announcements

Script after hours

»



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Time Conditions

Used as Destination by 1 Object (Click to Expand) ▼

Edit Time Condition: Holidays (*272)

Time Condition name ⓘ

Override Code Pin ⓘ

Invert BLF Hint ⓘ

Change Override ⓘ
Current: Unknown State

Time Zone: ⓘ

Mode ⓘ

Time Group ⓘ

Destination matches ⓘ

Announcements

script_HOLIDAYS

Destination non-matches ⓘ

Time Conditions

Business Hours

»

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Task 8

Complete the post-installation checklist in appendix 'A'. Use today's date for acceptance and cutover dates.

Submission requirements

You are required to submit the following as evidence for this assessment:

1. Written tasks should be completed on a word processor
2. Screen shots for each component of your system build.
3. Screen shots for each functionality test
4. Voice recordings you made for your IVR as .wav files
5. Completed appendix A
6. You must click the submit button