

We VoIP Service for the Samsung Communication Manager v4.0.0.8

WE-VoIP Client V3.5.0.2



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2. INTRODUCTION

The purpose of this manual is to introduce the Samsung WE VoIP Client for Smart Phones and the programming procedures to set up WE VoIP service for the Samsung Communication Manager (SCM) over an industry standard Wi-Fi network.

This mobile SIP client application makes your smart phone a mobile extension of the SCM phone system. With WE VoIP, you can make or answer VoIP calls using the default dialer and contacts of your smartphone. Each WE-VoIP extension must have a license. The Samsung Mobile License includes a separate field that determines the number of WE-VoIP clients.



WE VoIP supports HD Voice. With HD Voice, callers can enjoy high-quality voice call during a VoIP call. While an HD Voice call is in progress, the HD Voice logo displays on the call screen.

3. WE-VOIP CLIENT INSTALLATION

This section covers smart phone hardware requirements, supported smart phone models, where to get the client software and how to load it on the smart phone.

Supported Devices

Client Type	Model	Android Version.
Samsung Galaxy Edition	Galaxy S3 Series	4.0.3 or higher
	Galaxy S4 Series	4.1 or higher
	Galaxy Note2	4.0.3 or higher

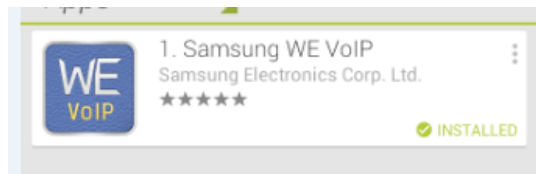
Continue to next page.

Download from Google Play Store



Access Play Store and search 'Samsung VoIP'.

This application will appear in the list. Select it.



Then click 'Install' button to begin download and installation.

It is recommended to move the application icon to your home screen for quick access to change settings.

End of Client application installation.

Stop at this point because the client will not register to the SCM until after the system WE VoIP programming options are completed. See section 5 of this document.

Changing the settings of the WE-VoIP Client is covered in section 6 of this document.

4. FEATURES LIST

The following list of features is available to WE-VoIP extensions registered to the SCM phone system. **Limitations:**

1. Service is only available on **SIP** and **PRI** trunks
2. SCM system software must be **v4.0.0.8 or higher**.

	FEATURE	SCM v4.0
1	Connect as a mobile SIP extension	Yes
2	Can be paired or stand-alone extension	Yes
3	Make & receive internal VoIP calls over Wi-Fi	Yes
4	Make & receive outside VoIP calls over Wi-Fi	Yes
5	Use default dialer of smart phone	Yes
6	Use Contact list of smart phone	Yes
7	VoIP calls share Call Logs of smart phone	Yes
8	Call Hold / Resume	Yes
9	Call Transfer (Blind transfer only)	Yes
10	Directed Call pickup of ringing extension	Yes
11	Group Pickup	Yes
12	Hold Pick Up (12 + XXXX)	Yes
13	Make station group calls	Yes
14	Set and Cancel DND using feature code	Yes
15	VM Message indication (mail icon upper line)	Yes
16	Missed Call Indication (Not for group calls)	Yes
17	Move VoIP call from Desk phone to mobile	Yes
18	Move VoIP call from mobile to desk phone	Yes
19	Forward when unregistered to Mobile number	Yes
20	Single CID number for paired WE VoIP ext.	Yes
21	Register to private IP address of the system	Yes
22	Register to public IP address of the system	Yes
23	WE-VoIP over LTE service	Yes
24	Manual Handover to Cellular network 'to Mobile'	Yes
25	Smart Routing >switch mobile number to WE VoIP	Yes
26	Auto updating of Client software by Google Play Store	Yes
27	Log gathering to assist technical support	Yes
28	Add Call (3 party conference call)	Yes
29	Hold On & Later > call answering options	Yes

HARDWARE

No New Hardware required for WE VoIP service	No
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5. PROGRAMMING PROCEDURES FOR WE VOIP SERVICE

This chapter lists programming procedures in SCM v4.0 or higher, required to set up WE-VoIP service. Each procedure is broken down sections corresponding to the SCM sections:

- General Description
 - This section will describe the purpose of this procedure.
- Programming
 - This section will detail any relevant SCM GUI changes relating to WE VoIP service.
- User Instructions (when applicable)
 - For features that are user-facing this section will describe how a user can access and use the feature

Notice

This section is designed with the understanding that the SCM system is already installed, programmed and operational. This means the SIP/PRI trunks, stations and Voice Mail are set up and functioning. Knowing the system was already fully operational will limit any potential trouble shooting to only WE VoIP service instead of general system setup.

Sample screen captures used in this document are from an SCM system running on release SCM v4.0. When programming a future software release the system they may appear slightly different.

TIP:

Samsung engineering recommends following these procedures in this order as the most efficient method.

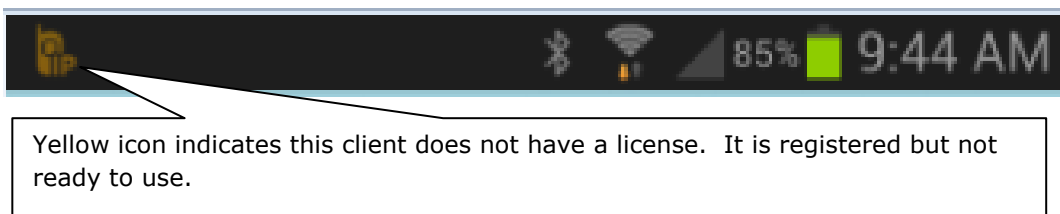
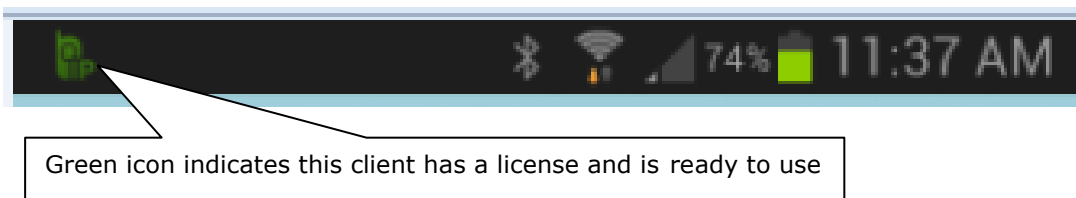
5.1 Enter WE-VoIP License

GENERAL DESCRIPTION

Sites that want to add or start out with WE VoIP extensions must order them as part of the SCM User License. (Samsung Mobile Phone)

WE VoIP user license is for concurrent users. So when 20 Samsung Mobile Phones are setup in the SCM system and Samsung Mobile Phone count is 20, only 20 WE VoIP users can be registered simultaneously.

When the 21st WE VoIP Client registers with the latest profile, he/she will see a **yellow** WE VoIP icon at the top of the phone instead of a green icon. Other clients exiting the WE-VoIP application, or turning off their phones will release licenses. This will take a minute or two, and then the yellow icon will turn green, indicating the Client is ready to make or receive calls. The user may also periodically press the Application icon to poll the system for an available license.



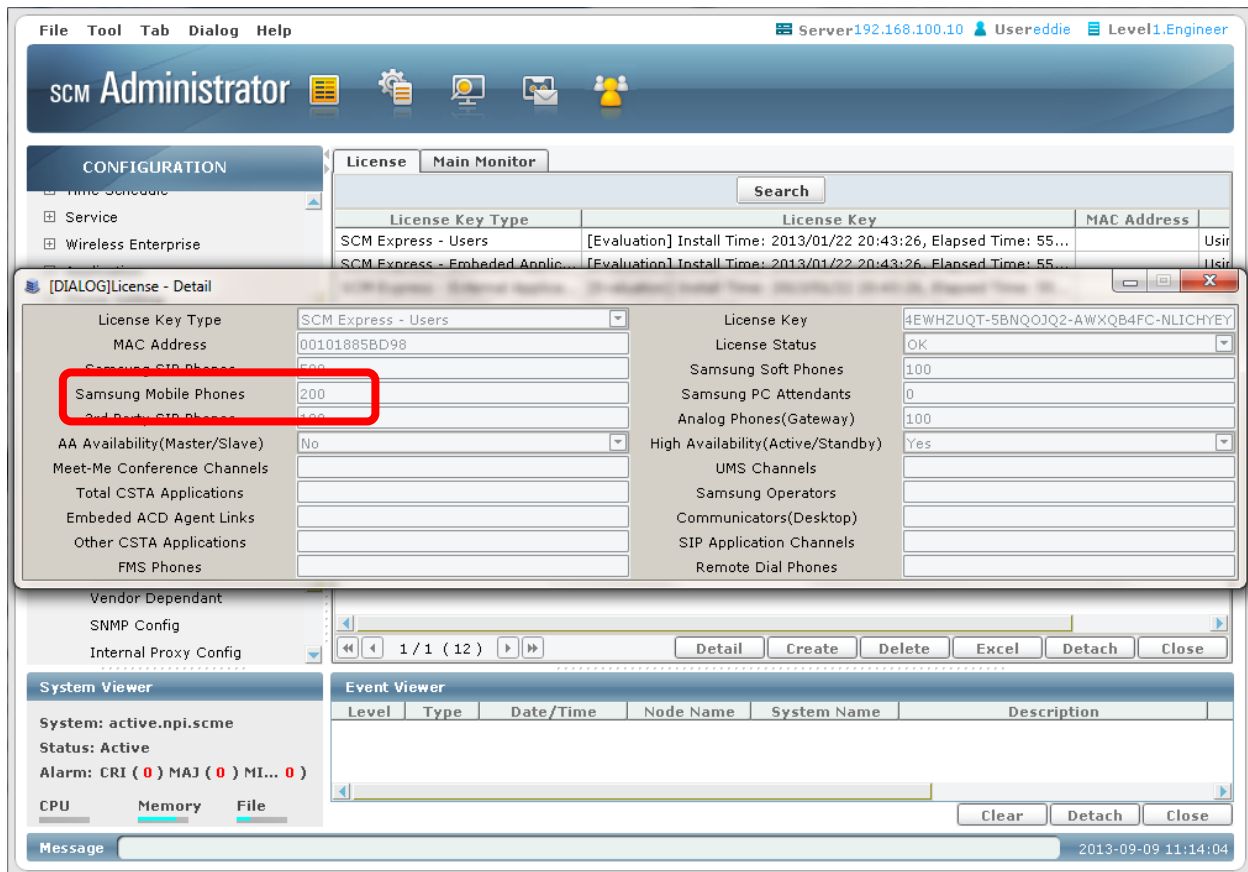
PROGRAMMING

WE VoIP Users in SCM User License

Please note: If you already have a User License in place, you will need to delete the license first before you can add your new User License that includes Samsung Mobile Phone count.

Upon receipt of the email with the SCM User License, select 'Create' then paste it into the field. Next, double click the license added and look to see your Samsung Mobile Phone count has been updated.

Here you will see that the license has been updated for Samsung Mobile Phones



Note: The User Evaluation License will support a maximum of 100 Samsung Mobile Phones for the 30 day evaluation period.

5.2 Mobile Service Options “Wi-Fi Access Point Added”

GENERAL DESCRIPTION

1. SCM will need to be told the SSID of the Wi-Fi network that the SCM system will be connected to. The WE-VoIP Client will connect to this wireless network before it can register to the IP address of the SCM system.

In order to do that, we need to add the SSID to the Mobile Service Options menu.

PROGRAMMING

The screenshot shows the 'Mobile Service Options - Change' configuration window. The 'SSID' field is highlighted with a red box and contains the text 'NPI Lab'. The window is divided into several sections:

- User Group:** dallas
- Remote Dial Public IP Address:** (empty)
- Mobile DISA Number:** (empty)
- Mobile VMS DISA Number:** (empty)
- WE Work Server IP Address:** (empty)
- WE Work Server Public IP Address:** (empty)
- WE VoIP CID Server IP Address:** (empty)
- WE VoIP CID Server Public IP:** (empty)
- WE Work Server Protocol:** HTTP
- WE VoIP CID Server Public Protocol:** HTTP
- Wait Call, Later Call:** True
- Auto Answer CLI Number:** (empty)
- Use 3G Call Only:** No
- Remote Dial Public Port:** (empty)
- Mobile DISA Code:** (empty)
- WE Work Server Port:** 80
- WE Work Server Public Port:** 80
- WE VoIP CID Server Port:** 80
- WE VoIP CID Server Public Port:** 80
- WE VoIP CID Server Protocol:** HTTP
- WiFi Band:** Auto
- Auto Answer Profile Number:** (empty)
- 3G Call Prefix:** (empty)
- 2.4G Channel List:**
 - CH 1: ☒ CH 2: ☐ CH 3: ☐ CH 4: ☐
 - CH 5: ☐ CH 6: ☒ CH 7: ☐ CH 8: ☐
 - CH 9: ☐ CH 10: ☐ CH 11: ☒ CH 12: ☐
 - CH 13: ☐
- 5G Channel List:**
 - CH 36: ☒ CH 40: ☒ CH 44: ☒ CH 48: ☒
 - CH 149: ☒ CH 153: ☒ CH 157: ☒ CH 161: ☒
 - CH 165: ☒

Buttons at the bottom: Change, Apply, Close.

Mobile Service Options “FUTURE RELEASE”

- **Wifi Band:** To reduce handover delay, select the channel scanning option that the Access Points can meet. **Default is Auto**
 - a. **Auto** – access points supports both 2.4 GHz & 5 GHz radios and all roaming channels are saved to Login Profile.
 - b. **2.4G Only** - access points supports only a 2.4 GHz radio and only 2.4G Roaming Scan Channels are saved to Login Profile.
 - c. **5G Only** - access points supports only 5GHz radio and only 5G Roaming Scan Channels are saved to Login Profile.

- Auto Answer CLI Number:
- Auto Answer Profile Number:
- Use 3G Call Only:
- 3G Call Prefix:
- Remote Dial Public IP Address
- Remote Dial Public Port
- Mobile DISA Number
- Mobile DISA Code
- Mobile VMS DISA Number

The following items require you use Samsung's Wireless LAN product.

SCM is optimized to work with Samsung WLAN.

WE Work Server IP Address		WE Work Server Port	80
WE Work Server Public IP Address		WE Work Server Public Port	80
WE VoIP CID Server IP Address		WE VoIP CID Server Port	80
WE VoIP CID Server Public IP		WE VoIP CID Server Public Port	80
WE Work Server Protocol	HTTP	WE VoIP CID Server Protocol	HTTP
WE VoIP CID Server Public Protocol	HTTP		

2.4G Channel List			
<input checked="" type="checkbox"/> CH 1	<input type="checkbox"/> CH 2	<input type="checkbox"/> CH 3	<input type="checkbox"/> CH 4
<input type="checkbox"/> CH 5	<input checked="" type="checkbox"/> CH 6	<input type="checkbox"/> CH 7	<input type="checkbox"/> CH 8
<input type="checkbox"/> CH 9	<input type="checkbox"/> CH 10	<input checked="" type="checkbox"/> CH 11	<input type="checkbox"/> CH 12
<input type="checkbox"/> CH 13			
			<input type="checkbox"/> Selected All
5G Channel List			
<input checked="" type="checkbox"/> CH 36	<input checked="" type="checkbox"/> CH 40	<input checked="" type="checkbox"/> CH 44	<input checked="" type="checkbox"/> CH 48
<input checked="" type="checkbox"/> CH 149	<input checked="" type="checkbox"/> CH 153	<input checked="" type="checkbox"/> CH 157	<input checked="" type="checkbox"/> CH 161
<input checked="" type="checkbox"/> CH 165			
			<input checked="" type="checkbox"/> Selected All

5.3 Defining a Single Phone User for WeVoip Client

GENERAL DESCRIPTION

Once you have added your licensing, you will want to select what extension/s will be used for the WeVoip Client/s.

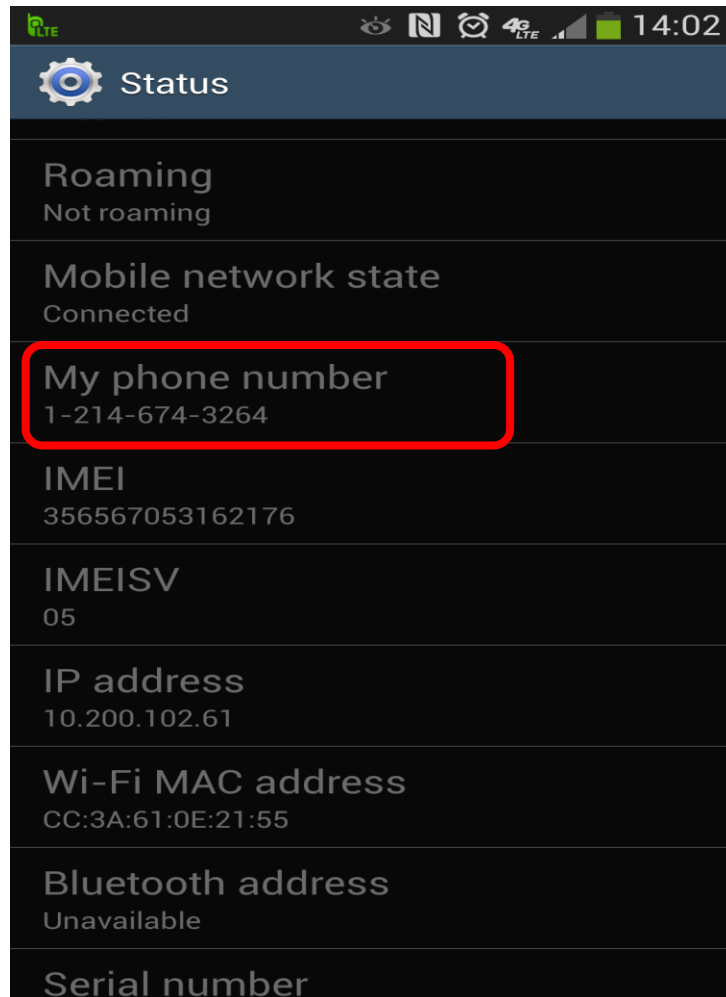
1. Go to single phone user and change the phone type to Samsung Mobile Phone.
2. Enter your phone number in the 'Mobile Phone Number' field. Set Phone Verification to MAC Address and enter the MAC address of your cell phone.

For reference please see figure below.

[DIALOG] Single Phone User - Change

User Group	dallas	Service Group	dallas-SG1
Location	dallas-LOC1	Extension Number	2177
Application User ID	2177@dallas.com	Extension Name	John Doe
Application Password	*****	Phone Verification	MACAddress
Authentication User ID	2177	MAC Address	5C:0A:5B:1A:CC:79
Authentication Password	****	Phone Type	Samsung-Mobile-Phone
IP Address		Language	English
Profile Login ID	2177	Use Mobile Phone Number	None
Profile Login Passcode		Media	RTP
Mobile Phone Number	19724895738	Ping Ring Type	Audio+Visual
TLS Connection	Normal	A-A Dual Registration	Enable
A-A Primary Node	NODE 0	Make Mailbox	Yes
VMS Extension Number		DTMF	RFC2833
URI Type	SIP	Time Zone	GMT -06:00 America/Chicago
RFC2833 DTMF Payload	101	Position	
Department		Service Group Local CLI Number	
Send CLI Number	8316487694	Restriction Policy	
Service Group Local Number		Gateway Name	
Class of Service		LDAP DN Number	
Extension Lock	None	Auto Answer by Click to Dial	Enable
Account Code Use	None	External Ringback Tone Use	None
Accept Login Override	Disable	Display Option	Normal
MOH Announcement ID		Call Monitoring	Disable
Send CLI Name		Use Virtual Ringback	Disable
Send Extension Number		Off Hook Alarm	
Caller Ring Type	None	MOH SIP Media Mode	Send Only
Check Registration Protocol	Disable	CMS Monitoring	Disable
Application Server Service Group			

Please Note: Make sure to get your cell phone number from the About Phone / About Device → Status screen. Some carriers have a “1” and some don’t, this matters when connecting your WeVoip Client to SCM



5.4 Pairing a Desk phone and WeVoip Client

GENERAL DESCRIPTION

Here we will pair our WeVoip Extension and our Desk phone.

Sample use cases

Inbound Call

- When an inbound call is sent to the desk phone, both the desk phone and the WE VoIP Client will ring.
- When the user is on an incoming call on the desk phone, he will be able to move the call to their WeVoip client using the move to mobile feature.
- If the user is on an inbound call on the WeVoip Client, he will be able to move the call to their desk phone using the mobile pickup feature.

Outbound Call

- When the user makes an outbound call on the desk phone, he will be able to move the call to their WeVoip client using the move to mobile feature.
- When the user makes an outbound call on the WeVoip Client, he will be able to move the call to their desk phone using the mobile pickup feature.

Example Setup: EXT 2017 (Desk phone) and EXT 2173 (WE VoIP Client)

1. Go to the EXT 2017 and setup the "Mobile Phone Number", we will enter in the WeVoip Client EXT 2173
2. Also, make sure that "Use Mobile Number" is set to **Both**

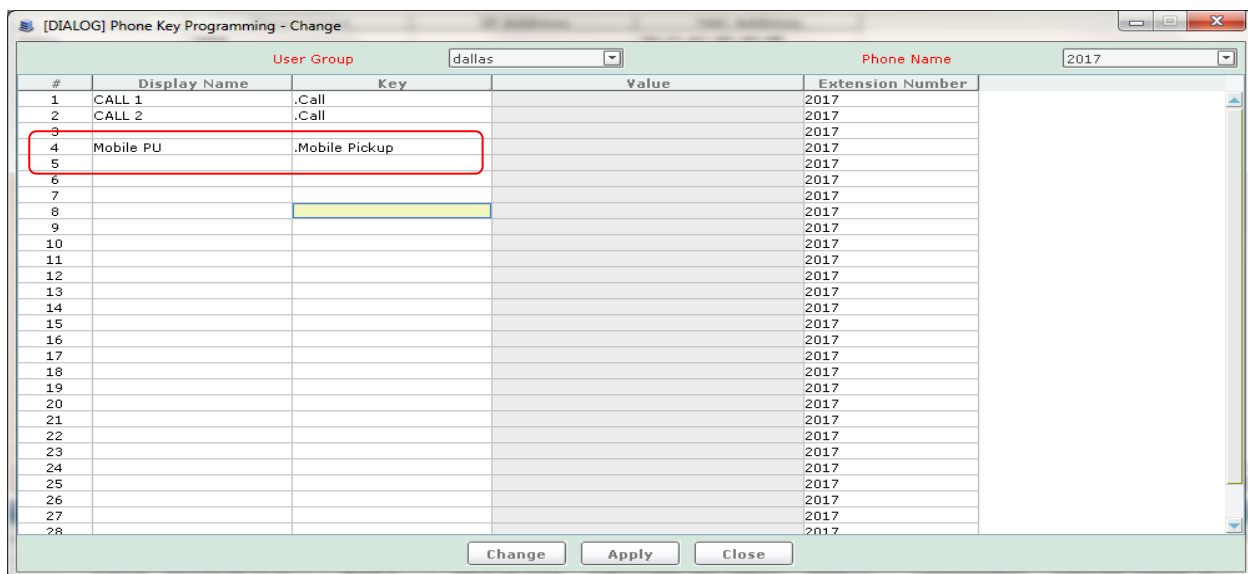
[DIALOG] Single Phone User - Change

User Group	dallas	Service Group	dallas-SG1
Location	dallas-LOC1	Extension Number	2017
Application User ID	2017@dallas.com	Extension Name	Eddie
Application Password	*****	PIN Number	****
Authentication User ID	2017	Phone Verification	MACAddress
Authentication Password	****	MAC Address	00:21:4C:97:19:84
IP Address	192.168.100.116	Private IP Address	192.168.100.116
Profile Login ID	5172011	Phone Type	Samsung-Desktop-Phone
Profile Login Passcode	****	Language	English
Mobile Phone Number	2173	Use Mobile Phone Number	Both
Protocol	UDP	Media	RTP
TLS Connection	Reuse	Ping Ring Type	Audio+Visual
A-A Primary Node	NODE 0	A-A Dual Registration	Enable
VMS Extension Number		Make Mailbox	Yes
URI Type	SIP	DTMF	RFC2833
RFC2833 DTMF Payload	101	Time Zone	GMT -06:00 America/Chicago
Department		Position	
Send CLI Number	8316487695	Service Group Local CLI Number	
Service Group Local Number		Restriction Policy	
Class of Service		Gateway Name	dallas-GW-2016
Extension Lock	None	LDAP DN Number	
Account Code Use	None	Auto Answer by Click to Dial	Enable
Accept Login Override	Disable	External Ringback Tone Use	None
MOH Announcement ID		Display Option	Normal
Send CLI Name		Call Monitoring	Disable
Send Extension Number		Use Virtual Ringback	Disable
Caller Ring Type	None	Off Hook Alarm	
Check Registration Protocol	Disable	MOH SIP Media Mode	Send Only
Application Server Service Group		CMS Monitoring	Disable

Change Apply Close

Moving a call in between devices

1. Only the SMT-I5230, SMT-I5243, SMT-I5210s, and SMT-I5220n phones support move to mobile, in this example the mobile number is our WeVoip.
2. To move a call from desk phone to WeVoip: while you are on your desk phone, go to the soft key menu → Func → up arrow a couple of clicks → Select move to mobile
3. In order to move a call from WeVoip back to the desk phone, you will need to have a Mobile Pickup Button added in phone key programming.



5.5 Multi-User with Deskphone and We VoIP

GENERAL DESCRIPTION

Here is this setup, the user will only have one EXT number, but that EXT will be shared on a desk phone and the WeVoip Client.

Sample Use Cases

Inbound Call

- If the user accepts an inbound call on the desk phone, he will be able to move the call to their WeVoip client using the move to multi feature.
- If an inbound call is sent to the desk phone, both the desk phone and the WeVoip Client will ring.
- You will only need to manage one mailbox

Outbound Call

- If the user makes an outbound call on the desk phone, he will be able to move the call to their WeVoip client using the move to multi feature.

Limitations

- Please note, at this time we do not support moving the call from WeVoip to desk phone in this multi-device setup.
- If you would like to do this, please see section 5.4

Example Setup: EXT 2075 will be used as a Multi-user & Multi-device

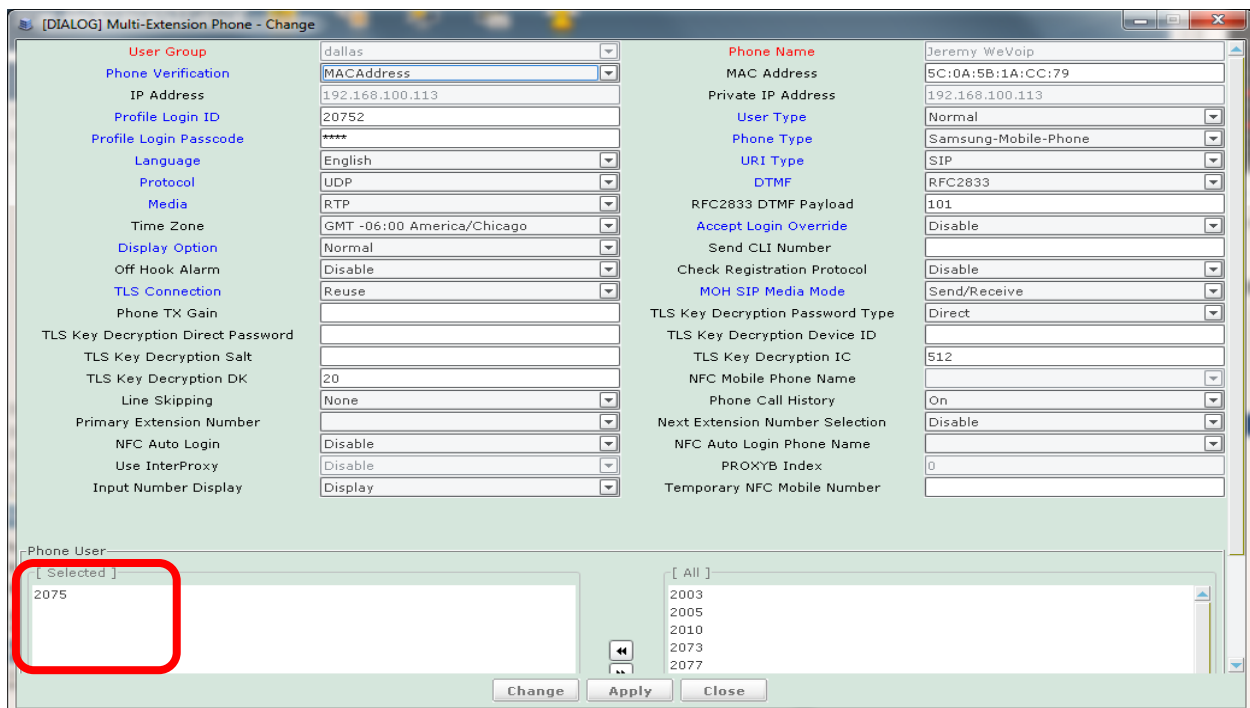
1. Create your Multi-Phone User
2. Also, be sure to select "Multi-Type" as Multi-Line & Multi Device
3. Mobile Phone Number = Your cell phone number

4. Now, create your first device "Multi-Extension Phone"
5. Be sure to move over the Multi-Phone User that you created previously

6. Then, create your second device "Multi-Extension Phone"
7. You will need to get the MAC Address of your cellphone first



8. Here you need to setup your WeVoip device, don't forget to add your user



Moving a call in between devices

1. In this setup, when we move a call from desk phone to WE Voip, we will use "Move to Multi".
2. This feature code must be defined in the menu for feature codes before it will appear on your desk phone.

Feature Service	Last Outgoing Redial	*31	1	30
Service Activation	Malicious Call Trace	*26	1	30
Class of Service	Meet Me Conference Join	*82	1	30
Feature Code	Mobile Pickup	*14	1	30
Activated Service List	Move to Mobile	*13	1	30
Service Permission	Move to Multi-Device	*132	1	30
	Move to Self Pickup			

3. To move a call from desk phone to WeVoip: while you are on your desk phone, go to the soft key menu → Func → up arrow a couple of clicks → Select move to multi → then press "OK"

5.6 Mobile Phone Profile

GENERAL DESCRIPTION

All WE VoIP Clients use the Mobile Phone Profile. Each time the application is started then stopped it will check for the download the latest Mobile Phone Profile. Also, if any changes are made to the use profile the phone will automatically update as long as it is connected to SCM.

Setting	Value
User Group	dallas
Mobile Phone Number	12145301234
Select Download Server	System
Roaming Trigger	-70
Roaming Scan Period	3
Noise Suppression TX	Disable
Echo Suppression	Enable
Enable Swing Free TX	Enable
Media Start Port	10000
Multiframe Enable	Disable
TOS Media Value(DSCP)	224
JBC Threshold	4
Extension Number	2075
User Agent Info	
Version	
Roaming Delta	10
Noise Suppression RX	Disable
AECM	Speaker Phone
Enable Swing Free RX	Enable
Enable CNG	Enable
Media End Port	30000
Multicast Enable	Disable
TOS Control Value(DSCP)	192

- **Mobile Number:** The mobile telephone number assigned to the WE-VoIP extension created in SIP numbering plan.
 - a. The telephone number must be exactly as it appears in **Settings-About Device -Status -My phone number**. Some carriers will use the "1" and some do not. WE VoIP will not work if this number is not entered exactly as it appears in your smart phone.
- **Select Download Server:** If external server is available, select that server number here.
- **Noise Suppression RX:** Select whether Noise Suppression is used or not in case of receiving an incoming call. (default: Disable)

When this option is enabled, Noise Suppression technology reduces stationary and transient noises in single-channel speech signals increasing the signal-to-noise ratio, improving speech intelligibility and reducing listening fatigue. (We recommend that you set this option as enable.)
- **Noise Suppression TX:** Select whether Noise Suppression is used or not in case of making an outgoing call. (Default: Disable)

When this option is enabled, Noise Suppression technology reduces stationary and transient noises in single-channel speech signals increasing the signal-to-noise ratio, improving speech intelligibility and reducing listening fatigue. (We recommend that you set this option as enable.)

- **AECM:** Select Auto Echo Cancellation Mode. (Default: Speaker Phone). Unless you are provided some special instruction, do not change default value.
- **Echo:** Select whether Echo Cancellation is used or not. (Default: Enable). Unless you are provided some special instruction, do not change default value.
- **Swing Free RX:** Select whether Diamond Voice solution is used or not. (Default: Enable). If this option is enabled, Diamond Voice will optimize the voice by following H/W specification. So voice quality will be increased when you receive an incoming call. Unless you are provided some special instruction, do not change default value.
- **Swing Free TX:** Select whether Diamond Voice solution is used or not. (Default: Enable). If this option is enabled, Diamond Voice will optimize the voice by following H/W specification. So voice quality will be increased when you receive an incoming call. Unless you are provided some special instruction, do not change default value.
- **CNG:** Select whether Comfort Noise Generator is used or not. (Default: Enable). If this option is enabled, CNG will generate comfort noise during the silence intervals, to avoid the “disconnected line” effect. Unless you are provided some special instruction, do not change default value.
- **Multi Frame:** Select whether Multi Frame of Voice engine is used or not. (Default: Disable). If this option is enabled, AP’s control ability will be increased and AP’s computing load will be reduced in case of connecting to Samsung AP/APC. We recommend that you set this option as disable for call stability.
- **Multicast:** Select whether Multicast of Voice engine is used or not. (Default: Disable). If this option is enabled, AP’s control ability will be increased and AP’s computing load will be reduced in case of connecting to Samsung AP/APC. We recommend that you set this option as disable for call stability.
- **TOS:** Set IP header TOS field for RTP media. Adjust TOS bits as required by the Network Administrator. (Default: 224)
- **Jitter Threshold:** Set Jitter Buffer size of the phone. (Default: 4). When Jitter Buffer is increased, delay is increased but the Jitter size is decreased. On the contrary, when Jitter Buffer is decreased, delay is decreased but the Jitter size is increased. In this case voice quality drops. Unless you are provided some special guide, do not change default value.

5.7 Access Code Solution “Local Calls”

GENERAL DESCRIPTION

When a WeVoip Client makes a trunk call, it will automatically use the default access code defined within the User Group menu.

There is one caveat

Most companies default access code is “9”. You will see this cause an issue if you dial a 972 number from your phone, SCM will remove the 9 in 9724443456, and only send out 724443456. This applies for any local area code that starts with the access code you are using.

THIS WILL BE FIXED IN FUTURE RELEASE

How to accommodate this

1. You will need to define your local area code as an access code, and set the type to internal.
2. In my example, 972 is my local number.
3. Make sure to set the digit length to equal the length of number dialed.
4. Please refer to Access Code Routing that you received training on in the SCM-Professional course for more routing info.

The screenshot displays the SCM configuration interface. On the left is a navigation tree with the following items: CONFIGURATION, Location, User Group, User, Trunk Routing, Route, Priority Routing, Location Based Routing, Access Code (highlighted), Time Based Routing, Load Balance Routing, and Common Route Prefix. The main panel has tabs for Priority Routing, Location Based Routing, Access Code, and Main Monitor. The 'Access Code' tab is active, showing a search area with 'User Group' and 'Access Number' dropdowns, and 'Search', 'Clear', and 'Reset' buttons. Below this is a table with the following data:

User Group	Access Number	Number Type	Location Based Routi...	Min Length	Max Length
dallas	9	Normal	NPI-Routing	1	40
dallas	972	Internal	NPI-Routing	11	11

Below the table is a 'DIALOG Access Code - Change' window. This dialog contains the following fields:

- User Group: dallas
- Access Number: 972
- Number Type: Internal
- Location Based Routing Name: NPI-Routing
- Minimum Digit Length: 11
- Maximum Digit Length: 11

At the bottom of the dialog are 'Change', 'Apply', and 'Close' buttons. A red rectangle highlights the 'User Group', 'Access Number', 'Number Type', and 'Location Based Routing Name' fields.

5.8 To Mobile Feature “Manual Handoff to Cell”

GENERAL DESCRIPTION

WeVoip supports a “To Mobile” function from the client. When a user presses the button ‘To Mobile’ on WE VoIP client, the VoIP call in progress will be redirected from its Wi-Fi network to the mobile network. Once engaged, the mobile phone will start ringing to establish the call. The current VoIP call will be placed on hold during the manual handover to the mobile network.

PROGRAMMING

As this time you should already have WeVoip setup on a phone and trunk routing setup correctly.

Go to Routing, the click change on the route used for outbound calls. Change the TIE Trunk field to “Tie”. This will enable to ability for the route to tie trunk calls together.

The screenshot shows the [DIALOG] Route - Change window with various configuration fields. The 'TIE Trunk' field is highlighted with a red rectangle and is set to 'Tie'. Other fields include Route Type, Route Name, Register Type, Port, Domain Name, Authentication Password, Outbound CLI Prefix, A-A Primary Node, Forced Send CLI Number, Send CLI Name for Inbound Call, Transfer Caller ID, Anonymous Call Reject, NAT Traversal, URI Type, Protocol Type, Registrar Address, Register Retry Interval(sec), Use Request URI User Info, Keep Alive Interval(sec), Keep Alive Retry Interval(sec), SIP P-Asserted-ID Type, Modify E.164 Format, Inbound Error Announcement, Blacklist Check Message, DNS SRV Version, DNS2, Failover Response, Retry Pause Time(sec), Maximum Call, User Group, Location, Proxy Server, User Name, Authentication User Name, DNS, DTS Mode, A-A Dual Registration, Send CLI Name for User, CLI for Forwarded Call, Anonymous URI, Tandem Diversion Number, Register Expires(sec), Maximum Register Retry, Call Authentication, Keep Alive, Maximum Keep Alive Retry, Keep Alive User Info, MOH SIP Media Mode, Outbound Error Announcement, Blacklist Expires(sec), DNS SRV Query, Secondary Proxy Server, Failover, Failover Timeout(sec), and Recovery Method.

5.9 Setup Call Forward Unreachable

GENERAL DESCRIPTION

When the WE VoIP SIP client is unreachable for the reasons listed below, the system can then forward the call to the **Call Forward Unreachable** destination entered in the unreachable field for that EXT number.

The unreachable destination is the SIP Extension associated with the WE-VoIP client in single phone user or your mutli-user, already covered in section 5.2 and 5.4 of this document.

- a. **Case:** – When a WE-Voip client is disconnected normally (unregistered), an incoming call to the client will be forwarded to the pre-assigned number in the Call Forward Unreachable field.

PROGRAMMING

In service activation for EXT 2173, we have setup our Preset Call Forward Unreachable Destination.

The screenshot displays the 'Service Activation - Activate' dialog box. The 'User Group' is set to 'dallas' and the 'Extension Number' is '2173'. The 'Service Type' is 'Preset Call Forward Unreachable'. The 'Destination' is '92146743264'. Other fields include 'No Answer Time(sec)' set to 10, 'Start Time', 'Service Date', 'Preset Call Forward Type' set to 'Both', 'Auto Record Call Type', 'Hot Line Delay(sec)', 'Service Schedule', 'Use Notification', 'End Time', 'Allow Other Ring', 'Auto Record Mailbox', 'Hot Desk Expire Time(hour)', and 'Incoming Call Logging'. The 'Service Activation' tab is selected in the background, showing a table of service types and their activation status.

Service Type	Service Activation	Value 1	Value 2
Call Forward All	Deactivated	-	-
Call Forward Busy	Deactivated	-	-
Call Forward No Answer	Deactivated	-	-
Call Forward Unreachable	Deactivated	-	-

5.10 Connecting WeVoip Client Remotely

GENERAL DESCRIPTION

This section will explain how to connect your WE VoIP client from an external Wi-Fi-network or over your 4G/LTE network.

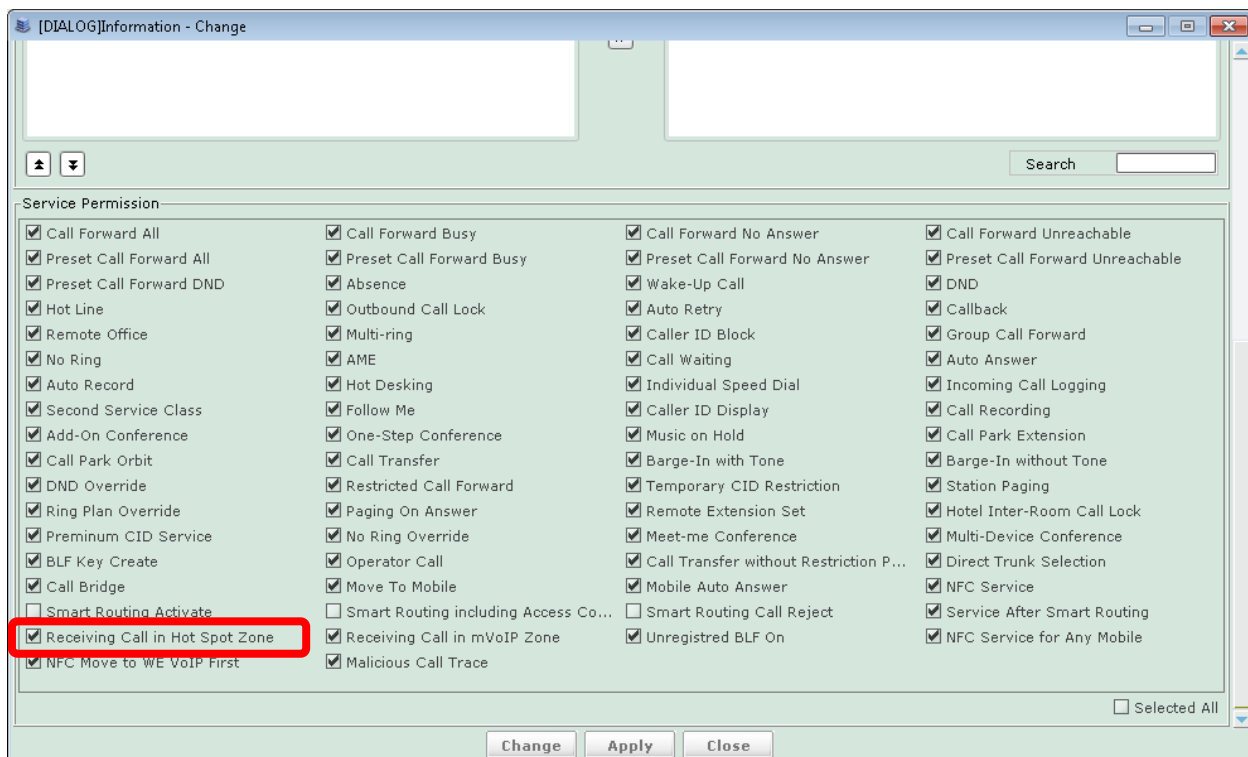
PLEASE NOTE

This will require you to have an SBC. Please refer to a separate SBC configuration document.

PROGRAMMING

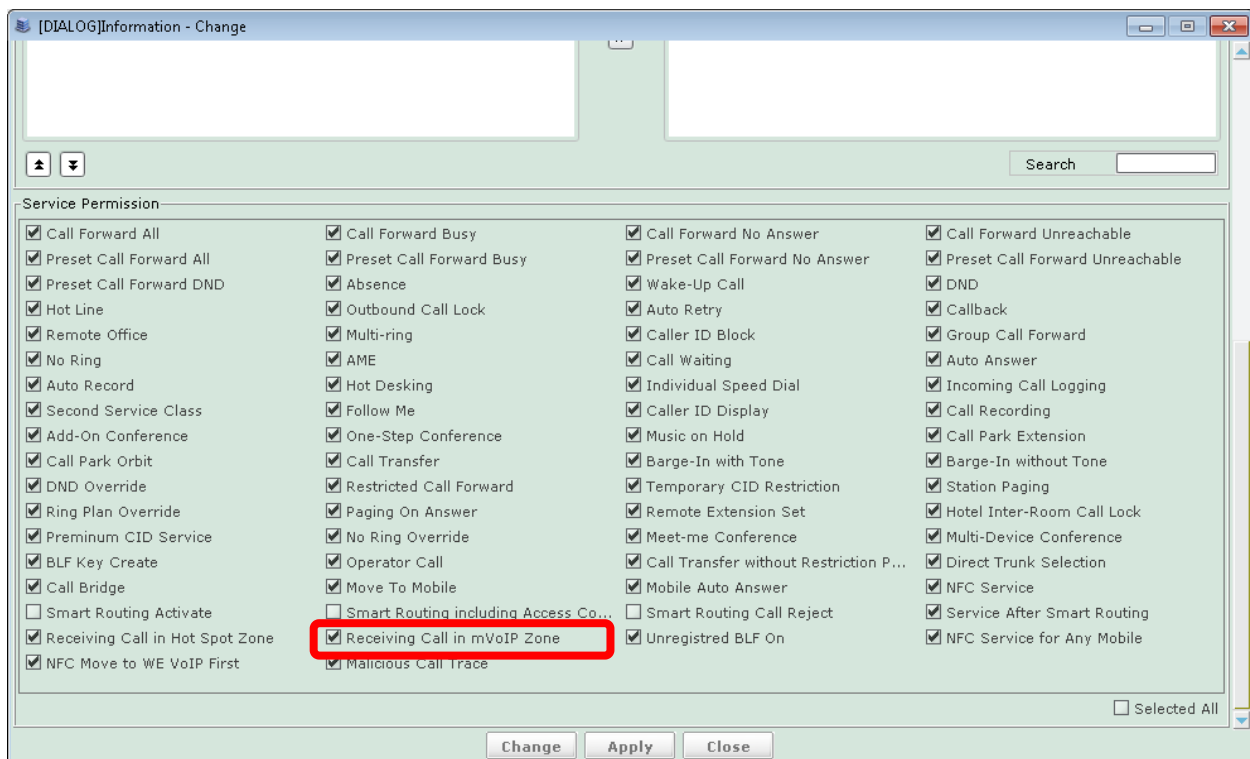
Receiving Call in Hot Spot Zone

This is used to setup the WeVoip subscribers to receive calls in the Hot Spot Zone. "Hot Spot Zone" means an external Wi-Fi network. We need to set the following items in the menu. [Configuration > User Group > Change User Group > Information]



Receiving Call in mVoIP Zone

This is used to setup the WeVoip subscribers to receive calls in the mVoIP Zone. "mVoIP Zone" is your data 4G/LTE network. We need to set the following items in the menu. [Configuration > User Group > Change User Group > Information]



6. USING THE WE VoIP CLIENT

6.1 Registering the WE VoIP Client to SCM

Registering the WE-VoIP Client to the Provisioning Server is the process for registering to the IP address of the SCM. This can be the Private or Public IP address as determined by the network administrator and company policy.

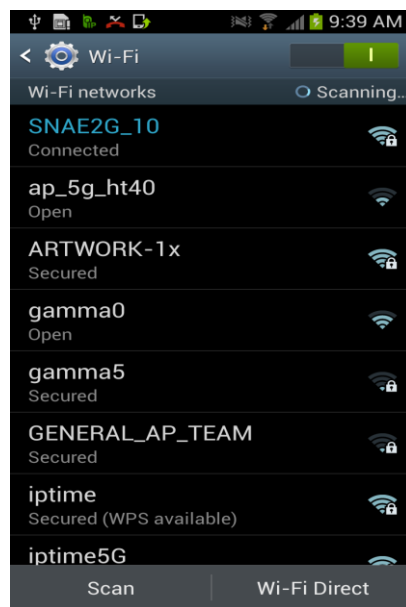
CHECKLIST

1. Before installing WE VoIP, you should update your smartphone to the latest firmware. If you are not using the latest firmware, you may experience poor sound quality during a call or other malfunctions.
2. Get the SSID of the wireless LAN the SCM is connected to.
3. Get the WE VoIP provision server information. This is the IP address of the SCM.

Note: The screen captures used in this section are from a Samsung Galaxy S3 from T-Mobile. These screens may appear slightly different from model to model, but the procedure and options are the same.

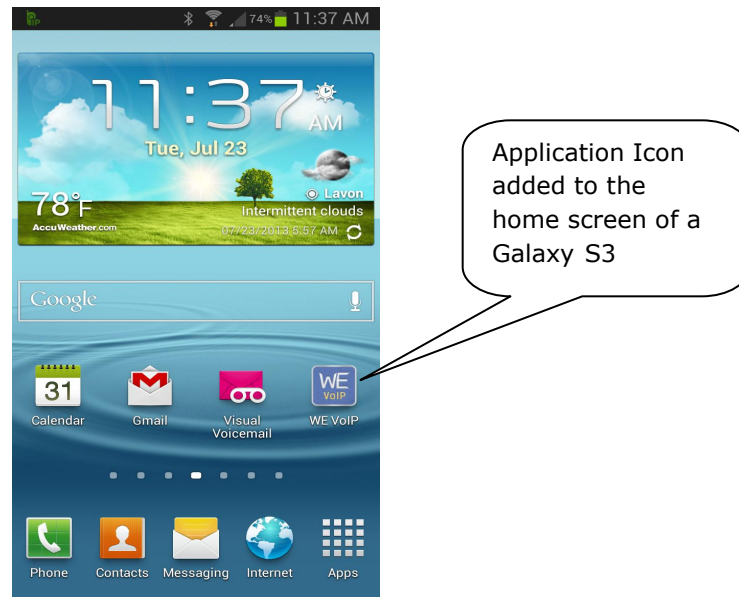
Step 1

Turn Wi-Fi on your smartphone and connect to the SSID assigned by the network administrator. This must be the same Wi-Fi Network the phone system is connected to. 'SNAE2G_10' illustrated on the picture below is an example of Wi-Fi network.



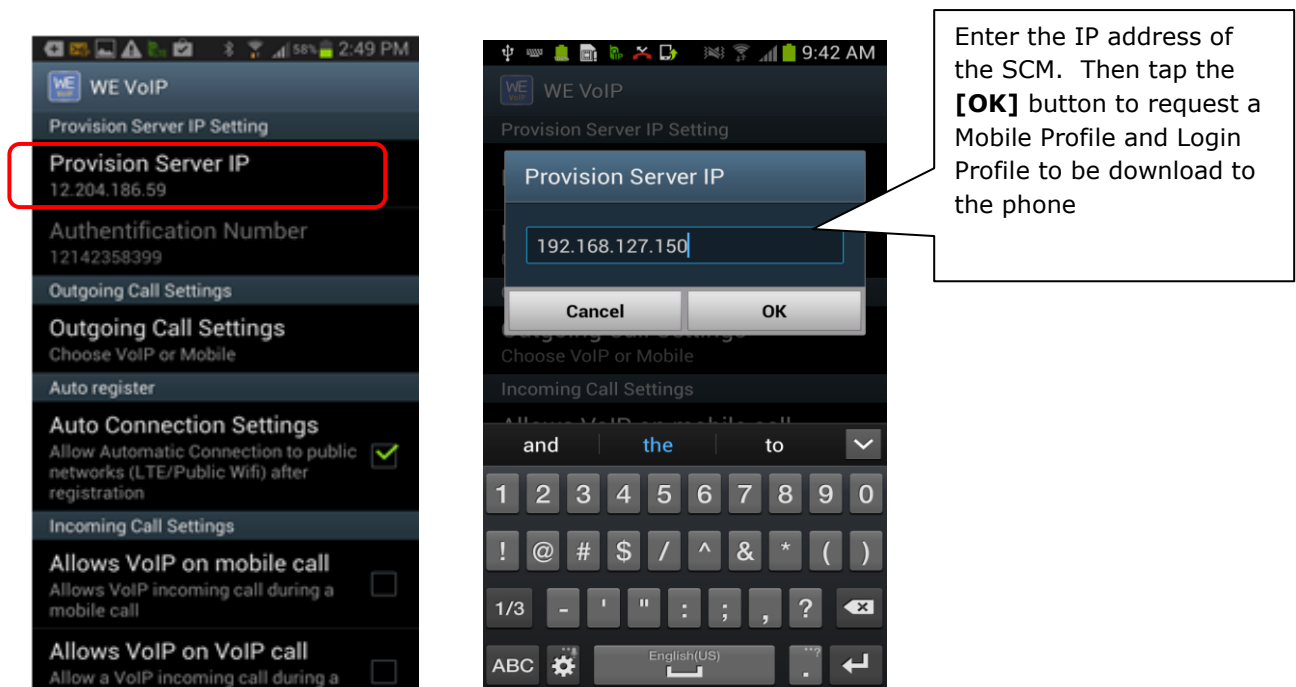
Step 2

The WE VoIP Client application should already be installed on the device as detailed in section 3 of this manual. Open the WE-VoIP application by tapping on the WE-VoIP Icon.




Step 3

On the application home screen tap on Provision Server IP and enter the IP address provided by the Network Administrator. It may be the private IP or the public IP, depending on where VoIP calling will be allowed.

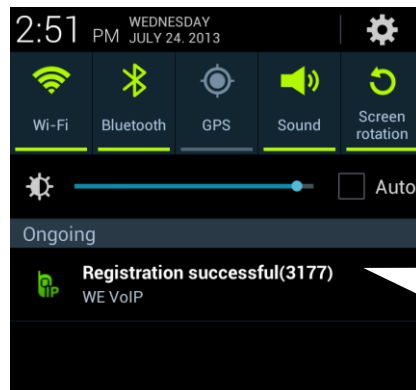


Step 4

When the profile is successfully connected, the icon indicating successful registration  appears at the top of the screen.

Drag top notification bar down to see the registration status.

If the registration fails, the failure icon  appears and the reason for the failure is displayed in a pop up window.



Registration is successful and displays the WE-VoIP extension number assigned to this client.

Green icon indicates successful registration



Red indicates unsuccessful registration

Red or Green icon will appear on the top line of the phone indicating the status of client application



Turning off Wi-Fi will cause icon to go red.



At this point the WE VoIP Client is ready to make and receive calls

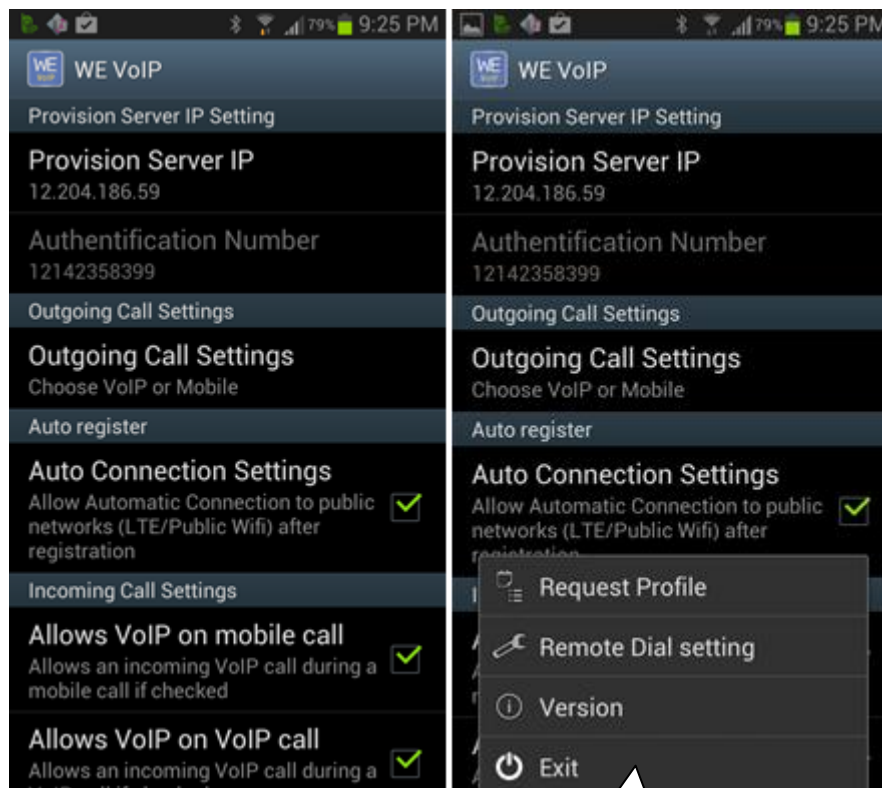
6.2 Client Main Menu Settings

USER SETTINGS

This chapter describes various settings/options and how to use. These are the same instructions in the same format that are in the WE VoIP User Guide for OfficeServ.



User Settings

[Tap] the WE VoIP application icon  to access the outgoing and incoming call settings required for using WE VoIP.



These options appear when you press the **MENU** button on the smartphone while the WE-VoIP application is open.

The following table explains the available settings in sequence as you scroll through them.

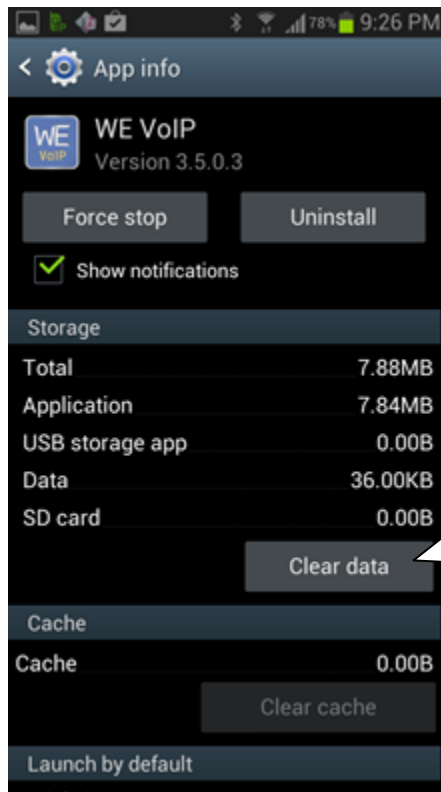
Menu	Description
Provision Server IP	You can enter the IP address of the provisioning server. This is the IP address of the phone system.
Authentication Number	This is your smartphone number. This is auto populated when you device uses a SIM card. The number must be exactly as it appears in your phone Settings > About phone If your provider does not use a SIM card this will be the Wi-Fi MAC address of your device
Outgoing Call Settings	<p>You can choose whether to use VoIP/4G or use 4G only for outgoing calls.</p> <ul style="list-style-type: none"> - Choose VoIP or Mobile: You will be prompted to select VoIP or 4G. - Use only Mobile: All outgoing calls are made over 4G network. <div>  <p>Even if Choose VoIP or Mobile is checked, outgoing calls are made over 4G network if you are not logged into WE VoIP.</p> </div>
Auto Connection Settings	Check this option to make the phone automatically register through public Wi-Fi / LTE after starting the application
Allow VoIP on mobile call	Allows an incoming VoIP call during a mobile call if checked
Allow VoIP on VoIP call	Allows a VoIP call while on another VoIP call.
Allow mobile call on VoIP	Allows an incoming mobile call during a VoIP call if checked.
Ringtone	<p>You can select a ringtone for an incoming WE VoIP call. Select [Default Ringtone] to use the same ringtone as the default ringtone of your smartphone.</p> <div>  <p>WE VoIP ringtone options are the same as the 4G ringtone options of your smartphone. If there is an incoming WE VoIP call when your smartphone is set to vibrate mode, your phone will vibrate without playing any ringtone.</p> </div>
Mute when Flipping	Mutes ringtone or vibration of incoming call by flipping your device in VoIP mode.
Do Not Disturb	Rejects an incoming call automatically
Auto Answer	You can choose whether to enable auto-answering when the switch To Mobile function is used. The switched Incoming mobile call is automatically answered.
Switching phones beep	Play beep sound when Auto answer switching phones
Beep when poor voice quality	This option will play a beeping sound when voice quality is poor.
Call alert failure levels	Select the level of poor quality that you want to be alerted to. This only works when the Beep when poor quality setting is selected.

Menu	Description
Update	You can use the WE VoIP update server to update the application. When the update file is downloaded successfully, the smartphone installation manager automatically starts to perform the application installation. If no updates are available, a popup message appears to notify that no updates are available.
Send log	You can send debugging log of the WE VoIP application to the server. This function is available when there is a log file created using the Write log function.
Write log	You can write a debugging log of the WE VoIP application. Turn this setting off to delete all previous logs. Log files are saved in :/storage/sdcard/smv
Premium CID Settings	This service is not available in North America
Show Context CID Information	This service is not available in North America.
mVoIP Settings	This service is not available in North America
Call Recording List	Tap this to access a list of the recorded calls you saved.
[Menu] → Request Profile	You can check for any changes in the profile, and if any, download the new profile from the server.
[Menu] → Remote Dial Setting	This service is not available in North America.
[Menu] → Version	You can view the version information of the WE VoIP application.
[Menu] → Exit	The WE VoIP application will be terminated.

Clear WE VoIP Application Data

When you need to clear all the settings and registration data go to:

MENU > settings > Application Manager > and TAP the WE VoIP application to the screen below.



Tap Clear Data to erase all the application data and profile data.

Then reenter the SCM IP address in the Provision Server IP setting.

You will need to request another profile after clearing this data.

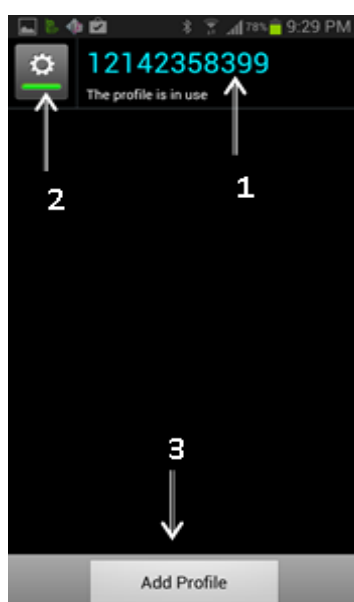
6.3 Administrator Settings


Administrator Settings

The Administrator Settings menu is provided only to the administrators for setting and controlling WE VoIP registrations. Normal users should not use the administrator settings menu unless it is absolutely necessary.

To access the Administrator Settings:

1. Click on the phone dialer to use the Dial Pad.
2. Dial [**1234##****] *Do not share this code with users.*
3. Press the green call button



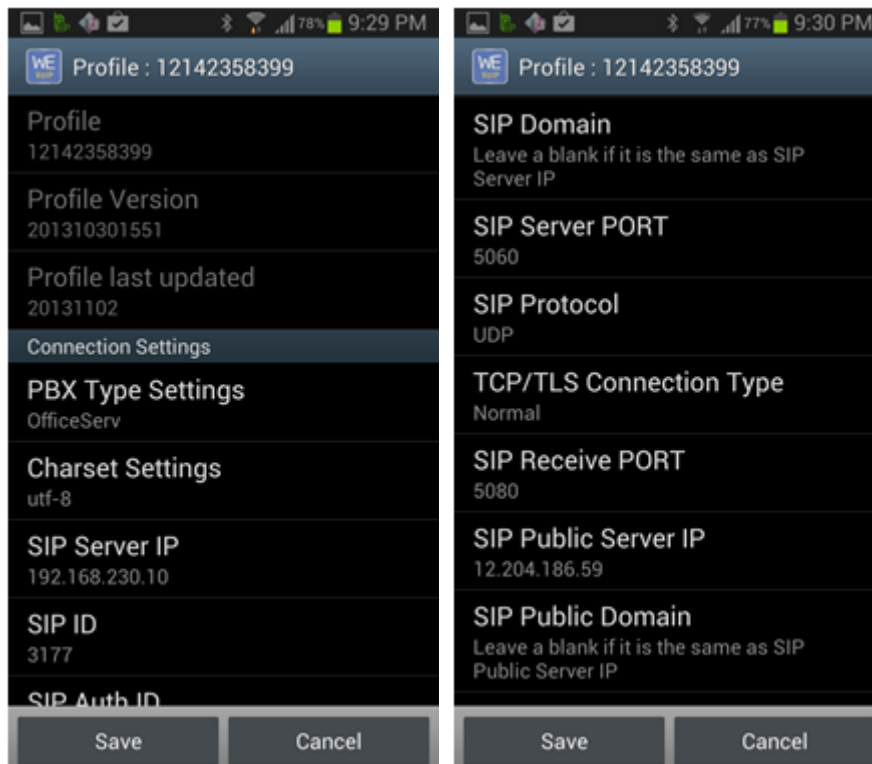
No.	Name	Function
①	Profiles	Name and usage status of the WE VoIP profile are displayed. Select this menu to open the edit profile screen. <u><i>You may have multiple profiles because your connect to other systems (branch office)</i></u>
②	Register profile button	This button registers the profile with the company's PBX and shows whether the profile is in use. Tap the button to attempt registering the profile. While the registration is being attempted, the registration button of the profile is shown as on () and the 'The profile is in use' message appears on the profiles list.
③	Add profile button	Tap the Add profile button to open the add and edit profile screen.

Add and Edit Profile Screen

You can configure various settings required for WE VoIP registration and usage by profile.


Connection Settings: allows you to set the WE VoIP registration;

General Settings: allows you to set voice options and other options.



Note: Continue scrolling down to see additional settings not shown in this manual.

After configuring the settings under each menu, tap the **[Save]** button to add the profile. (Settings items marked with * are required.)

Then, tap the  button on the left of the profiles list from the Administrator Settings screen to attempt the WE VoIP registration. If the profile you are saving is the first profile, its registration is automatically attempted when you save it.



Administrator Settings

Each settings items are related to the profile downloaded from the OfficeServ. Therefore, even if you have manually configured the settings, all previously configured settings are deleted when you download the profile from the OfficeServ. Profile download is performed automatically once everyday or when any changes are made to a profile.

The following tables list these available Profile settings.

Menu	Description
Profile	Shows the name of this profile. Read only
Profile Version	This is the profile version. Read only
Profile last updated	The date of this profile was last updated Read only
PBX Type Settings	You can set the type of the PBX that you want to register WE VoIP to. Default will be downloaded from mobile profile.
Charset Settings	Character settings
SIP Server IP*	You can enter the SIP server IP address of the PBX system.
SIP ID*	You can enter the SIP ID (phone number) of the PBX system.
SIP Auth ID	You can enter the SIP authentication ID of the PBX system. (Default: SIP ID)
SIP Auth PWD	You can enter the SIP server authentication password of the PBX system.
SIP Domain	Leave this blank if it is the same as SIP server IP. You can enter the SIP server domain address of the PBX system. (Default: SIP server IP)
SIP Server Port	You can enter the SIP server data receive port number of the PBX system. (Default: 5060.)
SIP Protocol	You can set the SIP protocol. The default and recommended setting is UDP.
TCP/TLS Connection Type	Set as Normal for OfficeServ
SIP Receive Port	You can enter WE VoIP local port number for receiving SIP data of the PBX system. (Default: 5080.)
SIP Public Server IP	You can enter the SIP server public IP address of the PBX system.
SIP Public Domain	You can enter the SIP public domain address of the PBX system. Leave this blank if it is the same as SIP Public Server IP.
SIP Public Server Port	You can enter the SIP server public port number of the PBX system. Default is 5060
Public TCP/TLS Connection Type Protocol	Set as Normal for OfficeServ
Add connection	You can set additional connections. This is Active-Active related settings of the SCM PBX system. You can configure the secondary SIP server information.
Add function	You can set additional functions. - MWI Feature Code: Set the function key of the MWI internal protocol. - Mobile Transfer: Enable the Mobile Transfer function. - Mobile Transfer Feature Code: Set the function key of Mobile Transfer. - Hold On/Later <i><u>Not supported on OfficeServ. Leave as Disabled</u></i> VM Transfer Feature Code Enter the OfficeServ Voice Mail group number to enable the Transfer to VM option for incoming calls.
Signal TOS	Set Signal TOS
RTP Media Port	Set port range for RTP media

Menu	Description
TLS Certificate setting type	Leave as > Use the default
Dial Rules	<p>You can configure prefix and Digit Map settings for making an external call.</p> <ul style="list-style-type: none"> - Prefix: Set the prefix code that will be used for making an external WE VoIP call. Generally this code is "9" (If you set a prefix, it is automatically added to the number of all outgoing external calls.) - Digit Map Rule: Set rules not to add a prefix. - Enable Digit Map Rule: If this is enabled, the prefix is automatically added. - Exception Rule: Set exception rule for making a call out of WE VoIP range. (An outgoing call is made over 4G network if the called number meets this rule.) - Enable Exception Rule: Allow exceptions.
Wi-Fi Settings	<p>You can configure Wi-Fi related settings.</p> <ul style="list-style-type: none"> - SSID: If you enter an SSID, WE VoIP registration is attempted only when the phone is connected to the specified SSID. - Roaming Trigger: Set Wi-Fi roaming parameters. - Wi-Fi Channel Country: Change the Wi-Fi country code for WE VoIP registration. - Wi-Fi Band: Set the Wi-Fi frequency band to scan for WE VoIP registration.
Audio Settings	<p>You can set codec and sound properties to use for a WE VoIP call.</p> <ul style="list-style-type: none"> - Codec Priority: Set audio codecs to use in a WE VoIP call and their priorities. - Sound Properties <ul style="list-style-type: none"> • Enable DV (Diamond Voice): Set whether to use DV filter of the WE VoIP application. • Swing Free Rx: Enable DV for Rx (reception). • Swing Free Tx: Enable DV for Tx (transmission). • CNG (Comfortable Noise Generation): Enable CNS. • TOS (Type Of Service): Set the TOS value.
DTMF Settings	<p>You can set the DTMF method during a WE VoIP call. <u>For OfficeServ always use rfc2833</u></p>
Security Settings	<p>You can set the security function available during a WE VoIP call.</p> <ul style="list-style-type: none"> - Enable Security: Enable the RTP security. - Enable AES: Enable Secure Realtime Transport Protocol (sRTP) Advanced Encryption Standard (AES). (This is automatically checked when Enable Security is selected.) - Use ARIA: Enable sRTP-AES/ARIA.
Auto Answer	You can choose to enable the auto answering function.
Auto answer number setting	You can enter a phone number to use for auto answering with the Remote Dial or Switch to Mobile function.
Corp logo URL setting	Settings to control Corporate logo
Choose Activity button String	Select language for WE VoIP and Mobile activity buttons
Remote Dial Settings	Ignore these settings. Not supported in North America
Premium CID settings	Ignore these settings. Not supported in North America

Menu	Description
Multiframe Mode	You can choose to enable Multiframe RTP Mode connecting with Samsung WE AP/APC.
Multiframe Silence Level	You can set the silence level for enabling the Multiframe RTP mode.
Multiframe Silence Sample	You can set the silence sample ratio for enabling Multiframe RTP mode.
MCS (MultiCall Simulator) Agent	You can choose to enable the interworking function with MCS equipment. (This setting is for engineers only and is independent of the profile.)
MCS Agent Settings	Ignore these settings for now.
Four Digit Calling	You can set a WE VoIP call to be automatically made when you press the extension number (4-digit). (This setting is for engineers only and is independent of the profile.)
Proximity Enable	You can choose to enable the proximity function. (This setting is for engineers only and is independent of the profile.)
Multi Device detail Setting	
Menu → Delete Profile	You can delete a profile. If you delete a profile, the profile will also be deleted from the administrator settings screen.



Dial Rules-DigitMap Settings

Refer to the following rules to set DigitMap.

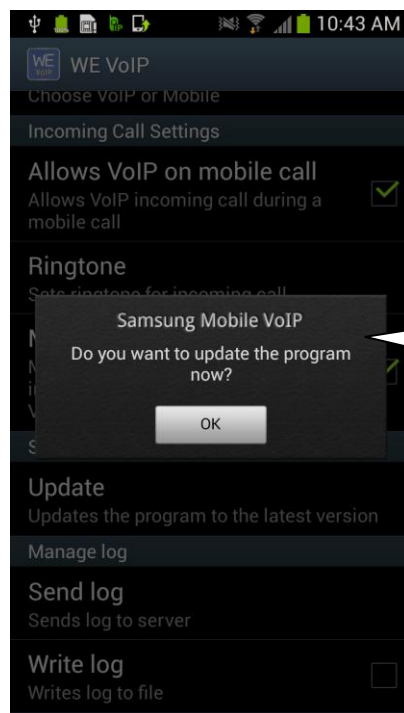
- XXXX: A prefix is not required for 4-digit numbers.
- #!: A prefix is not required for a number starting with #.

If the DigitMap is set as 'XXX, *!', you can dial a number starting with 3 digits + * without entering a prefix.

6.4 Update Client

Update Client Software

An alert popup appears when the program needs to be updated. Tap the **[OK]** button to start the update

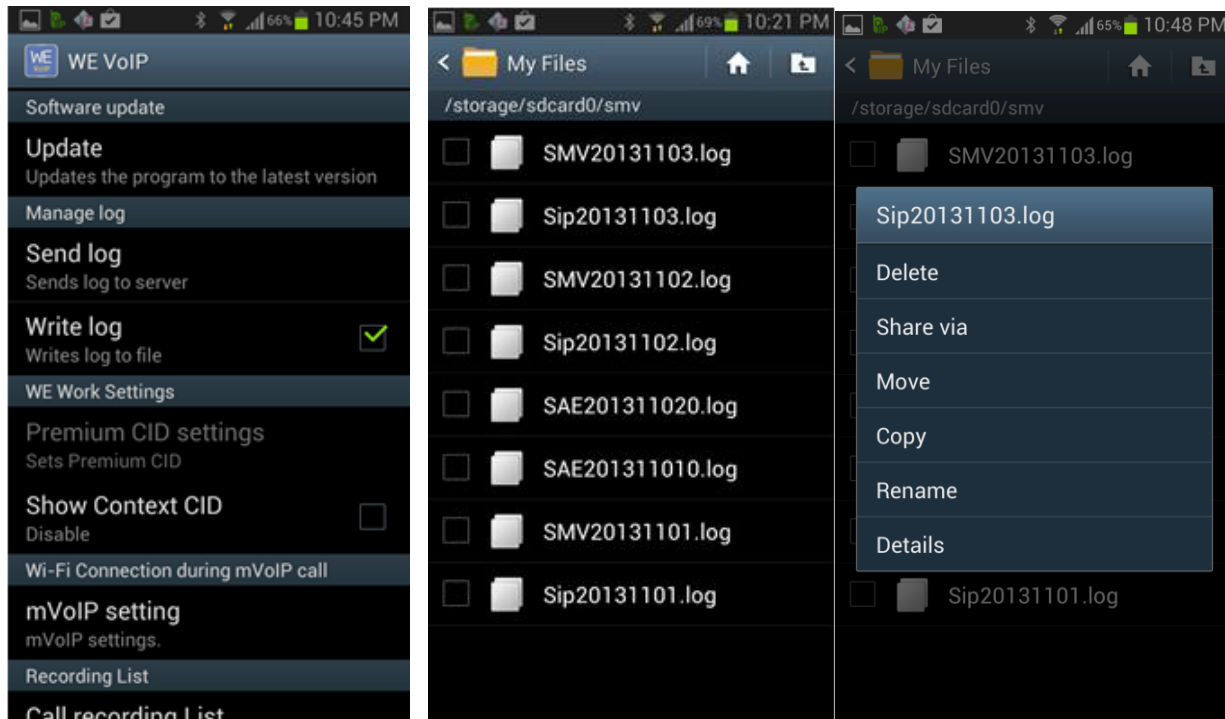


This message will only appear when the profile detects there is a newer version of the client on the upgrade server.

The upgrade server must be configured by the network administrator.

6.5 Trouble Shooting Logs

The Client application can write fault logs to a folder on your phones. These logs can be shared via email for review by technicians or engineers at Samsung Technical Support.



Instructions:

1. Check the Write log box.
2. Make the call or action that will duplicate the incorrect action.
3. Go to /storage/sdcard0/smv folder to access the log.
4. Momentarily tap and hold the selected log to get the share option.
5. Select the method to email the log and enter the email address to send the log to.

End of Document