We VoIP Service for the Samsung Communication Manager v4.0.0.8

WE-VoIP Client V3.5.0.2



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WE-VoIP Client for SCM v4.o.o.8

1. TABLE OF CONTENTS

1.	Table of Contents	1
2.	Introduction	2
3.	WE-VoIP Client Installation	3
4.	FEATURES LIST	5
5.	Programming Procedures for WE VoIP Service	6
	5.1 Enter WE-VoIP License	.7
	5.2 Mobile Service Options "Wi-Fi Access Point Added"	.9
	5.3 Defining a Single Phone User for WeVoip Client1	.1
	5.4 Paring a Desk phone and WeVoip Client1	.3
	5.5 Multi-User with Deskphone and We VoIP1	.6
	5.6 Mobile Phone Profile	20
	5.7 Access Code Solution "Local Calls"	22
	5.8 To Mobile Feature "Manual Handoff to Cell"2	23
	5.9 Setup Call Forward Unreachable2	24
	5.10 Connecting WeVoip Client Remotely2	25
6. U	sing the WE VoIP Client 2	27
	6.1 Registering the WE VoIP Client to SCM	27
	6.2 Client Main Menu Settings	30
	6.3 Administrator Settings	34
	6.4 Update Client	39
	6.5 Trouble Shooting Logs4	10

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.

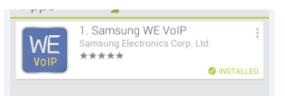
Download from Google Play Store



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

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)
 - $\circ~$ 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.

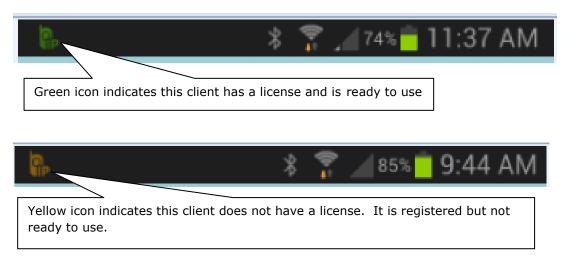
5.1 Enter WE-VoIP License

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.

WE-VoIP Client for SCM v4.o.o.8

File Tool Tab Dialog Help		🖴 Server192.16	58.100.10 👗 Usereddie 🛛 🗏 Level1.Engineer
SCM Administrator	📕 🏝 🖳 🖼	*	
	License Main Monitor	Search	
🗄 Service	License Key Type	License Key	MAC Address
🗄 Wireless Enterprise		[Evaluation] Install Time: 2013/01/22 20:43 [Evaluation] Install Time: 2013/01/22 20:43	
[DIALOG]License - Detail		LEVALUATION LIDSTAIL LIME: ZUL3/UL7ZZ ZU:43	
License Key Type	SCM Express - Users	License Key	4EWHZUQT-5BNQ0JQ2-AWXQB4FC-NLICHYEY
MAC Address	00101885BD98	License Status	ОК
Compare SID Desses	500	Samsung Soft Phones	100
Samsung Mobile Phones	200	Samsung PC Attendants	0
and Besty STR Bhenne	100	Analog Phones(Gateway)	100
AA Availability(Master/Slave)	No	 High Availability(Active/Standby) 	Yes
Meet-Me Conference Channels		UMS Channels	
Total CSTA Applications		Samsung Operators	
Embeded ACD Agent Links		Communicators(Desktop)	
Other CSTA Applications		SIP Application Channels	
FMS Phones		Remote Dial Phones	
Vendor Dependant			
SNMP Config			
Internal Proxy Config	✓ ((1/1 (12))))	Detail Create De	lete Excel Detach Close
System Viewer	Event Viewer		
System: active.npi.scme	Level Type Date/Tim	e Node Name System Name	Description
Status: Active			
Alarm: CRI (0) MAJ (0) MI (
CPU Memory File			Clear Detach Close
Message			2013-09-09 11:14:04

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

CONFIGURATION	[DIALOG]Mobile Service Options - Ch	ange				
Location	User Group	dallas		SSID	NPI Lab	
	Remote Dial Public IP Address			Remote Dial Public Port		
User Group	Mobile DISA Number			Mobile DISA Code		
🗄 User	Mobile VMS DISA Number					
Trunk Routing	WE Work Server IP Address			WE Work Server Port	80	
🗄 Time Schedule	WE Work Server Public IP Address			WE Work Server Public Port	80	
🗄 Service	WE VoIP CID Server IP Address			WE VoIP CID Server Port	80	
🖯 Wireless Enterprise	WE VoIP CID Server Public IP			WE VoIP CID Server Public Port	80	
Upgrade Mobile Software	WE Work Server Protocol	НТТР	•	WE VoIP CID Server Protocol	НТТР	-
Mobile Phone Profile	WE VoIP CID Server Public Protocol	НТТР				
Mobile Service Options	Wait Call, Later Call	True	-	WiFi Band	Auto	-
Mobile Configuration	Auto Answer CLI Number			Auto Answer Profile Number		
FMS Zone	Use 3G Call Only	No	•	3G Call Prefix		
🗄 Wifi Agent Configuration	2.4G Channel List					
APC List	✓ CH 1	CH 2		СН 3	СН 4	
Application	🗆 сн 5	🗹 CH 6		СН 7	СН 8	
	СН 9	CH 10		🗹 CH 11	🗌 CH 12	
Announcement	CH 13					
H Announcement						Selected All
System Viewer	-5G Channel List					
	✓ CH 36	🗹 CH 40		✓ CH 44	🗹 CH 48	
System: active.npi.scme	✓ CH 149 ✓ CH 165	🗹 CH 153		🗹 CH 157	🗹 CH 161	
Status: Active	CH 165					
Alarm: CRI (<mark>0</mark>) MAJ (<mark>0</mark>) MI						🗹 Selected All
CPU Memory File			Change Ap	ply Close		
CPU Memory File			Change Ap	ply 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 optomized to work with Samsung WLAN.

WE Work Server IP Address WE Work Server Public IP Address WE VoIP CID Server IP Address WE VoIP CID Server Public IP WE Work Server Protocol WE VoIP CID Server Public Protocol		WE Work Server Port WE Work Server Public Port WE VoIP CID Server Port WE VoIP CID Server Public Port WE VoIP CID Server Protocol	80 80 80 80 HTTP	
-2.4G Channel List	☐ CH 2 ✔ CH 6 ☐ CH 10	☐ CH 3 ☐ CH 7 ✔ CH 11	☐ CH 4 ☐ CH 8 ☐ CH 12	Selected All
-5G Channel List- ✓ CH 36 ✓ CH 149 ✓ CH 165	☑ CH 40 ☑ CH 153	☑ CH 44 ☑ CH 157	✔ CH 48 ✔ CH 161	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 extention/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 yopur cell phone.

For reference please see figure below.

😹 [DIALOG] Single Phone User - Cha	inge	_			. 🗆	X
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	****		During 1			
Authentication User ID	2177		Phone Verification	MACAddress		•
Authentication Password	****		MAC Address	5C:0A:5B:1A:CC:79		
IP Address						
Profile Login ID	2177		Phone Type	Samsung-Mobile-Phone		•
Prome Login Passcoue			Language	English		
Mobile Phone Number	19724895738		Use Mobile Phone Number	None		-
			Media	RTP		-
TLS Connection	Normal	-	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/Chicag	0	-
Department		•	Position			-
Send CLI Number	8316487694		Service Group Local CLI Number			-
Service Group Local Number			Restriction Policy			-
Class of Service		-	Gateway Name			-
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		-

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

PLTE .	& R	I 🗭	1	14:02
🔯 Status				
Roaming Not roaming				
Mobile network	state	j		
My phone numb 1-214-674-3264	er			
IMEI 356567053162176				
IMEISV ⁰⁵				
IP address 10.200.102.61				
Wi-Fi MAC addre	ess			
Bluetooth addres	SS			
Serial number				

5.4 Paring 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

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	LIDP	-	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		v	Gateway Name	dallas-GW-2016	
Extension Lock	None	v	LDAP DN Number		_
Account Code Use	None	v	Auto Answer by Click to Dial	Enable	
Accept Login Override	Disable	v	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	
pplication Server Service Group		-	CMS Monitoring	Disable	_

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.
- 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.

		User Group dallas	-	Phone Name	2017
¢	Display Name	Key	Value	Extension Number	
	CALL 1	.Call		2017	
2	CALL 2	.Call		2017	
-				2017	
ł	Mobile PU	.Mobile Pickup		2017	
5				2017	
)				2017	
,				2017	
3				2017	
)				2017	
0				2017	
1				2017	
2				2017	
3				2017	
4				2017	
5				2017	
6				2017	
7				2017	
3				2017	
Ð				2017	
0				2017	
1				2017	
2				2017	
3				2017	
4				2017	
5				2017	
5				2017	
7				2017	
3				2017	

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

	dallas	-	Service Group	dallas-SG1	
Location	dallas-LOC1	-	Extension Number	2075	
Application User ID	2075@dallas.com		Name	Jerenny G	
PIN Number	****		Mobile Phone Number	12145301234	
Use Mobile Phone Number	None	-	Department		
Position		-	Send CLI Number	9726522075	
Send CLI Name			Service Group Local Number		
ervice Group Local CLI Number		-	Multi Type	Multi Line & Multi Device	[
Call Appearance	MCA	-	Extension Lock	None	
Class of Service		-	Restriction Policy		[
Gateway Name		-	Authentication User ID	2075	
Authentication Password	****		MOH Announcement ID		[
Account Code Use	None	-	LDAP DN Number		
Auto Answer by Click to Dial	Enable	-	External Ringback Tone Use	None	[
Call Monitoring	Disable	-	Send Extension Number		
Use Virtual Ringback	Disable	_	Multi-Device Conference Join	Disable	[
Caller Ring Type	None	_	Application Server Service Group		
Ping Ring Type	None	-	CMS Monitoring	Disable	[
A-A Primary Node	NODE 0	-	A-A Dual Registration	Enable	[
VMS Extension Number			Call Recording Method		[
Allow Selective Call		-	Phone Display Name	Extension Number	[
Make Mailbox	Yes	-			

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

[DIALOG] Multi-Extension Phone - Change						
User Group	dallas	-	Phone Name	Jeremy 5210s		
Phone Verification	MACAddress	-	MAC Address	F4:D9:FB:1D:0D:50		
IP Address	192.168.100.130		Private IP Address	192.168.100.130		
Profile Login ID	20751		User Type	Normal		
Profile Login Passcode	****		Phone Type	Samsung-Desktop-Phone		
Language	English	-	URI Type	SIP		
Protocol	UDP	-	DTMF	RFC2833		
Media	RTP	-	RFC2833 DTMF Payload	101		
Time Zone	GMT -06:00 America/Chicago	-	Accept Login Override	Disable 💌		
Display Option	Normal	-	Send CLI Number			
Off Hook Alarm	Disable	-	Check Registration Protocol	Disable 💌		
TLS Connection	Reuse	-	MOH SIP Media Mode	Send/Receive		
Phone TX Gain			TLS Key Decryption Password Type	Direct 💌		
TLS Key Decryption Direct Password			TLS Key Decryption Device ID			
TLS Key Decryption Salt			TLS Key Decryption IC	512		
TLS Key Decryption DK	20		NFC Mobile Phone Name			
Line Skipping	None	-	Phone Call History	On 🔻		
Primary Extension Number		-	Next Extension Number Selection	Disable 💌		
NFC Auto Login	Disable	-	NFC Auto Login Phone Name			
Use InterProxy	Disable	-	PROXYB Index	0		
Input Number Display	Display	-	Temporary NFC Mobile Number			
Phone User						
	2073					
	Change		Close			

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

🕅 🐼 🔊 🧐 🚓 📶 🛑 14:02
🧔 Status
Roaming Not roaming
Mobile network state
My phone number 1-214-674-3264
IMEI 356567053162176
IMEISV ⁰⁵
IP address 10.200.102.61
Wi-Fi MAC address CC:3A:61:0E:21:55
Bluetooth address ^{Unavailable}
Serial number

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

[DIALOG] Multi-Extension Phone - Change					
User Group	dallas	-	Phone Name	Jeremy WeVoip	
Phone Verification	MACAddress	-	MAC Address	5C:0A:5B:1A:CC:79	
IP Address	192.168.100.113		Private IP Address	192.168.100.113	
Profile Login ID	20752		User Type	Normal	[
Profile Login Passcode	****		Phone Type	Samsung-Mobile-Phone	[
Language	English	-	URI Type	SIP	[
Protocol	UDP	-	DTMF	RFC2833	
Media	RTP	-	RFC2833 DTMF Payload	101	_
Time Zone	GMT -06:00 America/Chicago	-	Accept Login Override	Disable	
Display Option	Normal	-	Send CLI Number		_
Off Hook Alarm	Disable	-	Check Registration Protocol	Disable	
TLS Connection	Reuse	-	MOH SIP Media Mode	Send/Receive	
Phone TX Gain			TLS Key Decryption Password Type	Direct	
FLS Key Decryption Direct Password			TLS Key Decryption Device ID		-
TLS Key Decryption Salt			TLS Key Decryption IC	512	-
TLS Key Decryption DK	20		NFC Mobile Phone Name		
Line Skipping	None	-	Phone Call History	On	
Primary Extension Number		-	Next Extension Number Selection	Disable	
NFC Auto Login	Disable	-	NFC Auto Login Phone Name		
Use InterProxy	Disable	-	PROXYB Index	0	
Input Number Display	Display	-	Temporary NFC Mobile Number		-
[All] [Selected] 2075 2003 2005					
2010 2073 2077 2077 2077 2077 2077 2077					
	change	Ab	p17 01030		

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 Multi-Device	*132	1	30
Service Permission		101	-	
	i interest contributor			

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.

😹 [DIALOG]Mobile Phone Profile - Char	ige					
User Group	dallas	Extension Number	2075	~		
Mobile Phone Number	12145301234	User Agent Info				
Select Download Server	System 💌	Version				
Roaming Trigger	-70	Roaming Delta	10			
Roaming Scan Period	3	Noise Supression RX	Disable	~		
Noise Supression TX	Disable	AECM	Speaker Phone	~		
Echo Suppression	Enable	Enable Swing Free RX	Enable	~		
Enable Swing Free TX	Enable	Enable CNG	Enable	~		
Media Start Port	10000	Media End Port	30000			
Multiframe Enable	Disable	Multicast Enable	Disable	▼		
TOS Media Value(DSCP)	224	TOS Control Value(DSCP)	192			
JBC Threshold	4]				
	Change Apply Close					

- **Mobile Number:** The mobile telephone number assigned to the WE-VoIP extension created in SIP numbering plan.
 - a. The telephone number must be <u>exactly</u> 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.

Ш

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

in.

- 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.

CONFIGURATION		Priority Routing L	ocation Based Rou	ting Access Cod	le Main Monitor			
	A	User Grou	p	S	▼ earch Clear	Access Number Reset		
🗄 User Group		User Group	Access Number	Number Type	Location Based Routi.	Min Length	Max Length	
🗄 User		dallas	9	Normal	NPI-Routing	1	40	
Trunk Routing		dallas	972	Internal	NPI-Routing	11	11	
Route		IALOG]Access Code - Char	nge					
Priority Routing		User Group	dallas		v	Access Number	972	
Location Based Routing		Number Type	Internal			n Based Routing Nam		
Access Code		Minimum Digit Length				imum Digit Length	11	
Time Based Routing		Finishen Bigit Longa				annann brigie borigen	<u></u>	
Load Balance Routing				Cha	inge Apply C	lose		
Common Route Prefix								

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.

Route Type	User Group	-	User Group	dallas	
Route Name	SBC-NPI-Network		Location	dallas-LOC1	
Register Type	None	-	Proxy Server	172.30.110.1	
Port	5060		User Name	SBC-NPI	
Domain Name			Authentication User Name	SBC-NPI	
Authentication Password	SBC-NPI		DNS		
Outbound CLI Prefix			DTS Mode	Disable	
A-A Primary Node	NODE 0	-	A-A Dual Registration	Enable	
Forced Send CLI Number	None	-	Send CLI Name for User		
Send CLI Name for Inbound Call	None	-	CLI for Forwarded Call	Originator	
Transfer Caller ID	Transfer Party Number	-	Anonymous URI	Anonymous Invalid	
Anonymous Call Reject	None	~			
NAT Traversal	Disable	-	TIE Trunk	Tie	
URI Type	SIP	-	Tandem Diversion Number		
Protocol Type	UDP	-	Register Expires(sec)		
Registrar Address			Maximum Register Retry	1	
Register Retry Interval(sec)			Call Authentication		
Use Request URI User Info	Disable	-	Keep Alive	Disable	
Keep Alive Interval(sec)	35		Maximum Keep Alive Retry	1	
Keep Alive Retry Interval(sec)	35		Keep Alive User Info	Disable	
SIP P-Asserted-ID Type	None	T	MOH SIP Media Mode	Send/Receive	
Modify E.164 Format	No	-	Outbound Error Announcement	Enable	
Inbound Error Announcement	Enable	-	Blacklist Expires(sec)	600	
Blacklist Check Message	Register,Options,Invite	_	DNS SRV Query	Disable	
DNS SRV Version	Version2	T	Secondary Proxy Server		
DNS2			Failover	Disable	
Failover Response			Failover Timeout(sec)	5	
Retry Pause Time(sec)			Recovery Method	Registration	
Maximum Call			Maximum Inbound Call		

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.

CONFIGURATION		Service Activation	Mobile Phone Pr	ofile	Mobile Servic	e Options	FMS Zor	ne APC List	t 🗌 Main Mor	nitor
		User Group	dalla	s	-	Exten	sion Numbe	r 217	'3	-
🗄 Location				Search	Clear	Reset	:			
🗄 User Group		Service	Type	Servic	e Activation	¥alı	ie 1		Value 2	_
🗄 User		Call Forward All	1700	Deactiv		-		-	Tulue L	
🗄 Trunk Routing		Call Forward Busy		Deactiv	ated	-		-		
🗄 Time Schedule		Call Forward No Answer		Deactiv	ated	-		-		
⊟ Service		Call Forward Unreachab	le	Deactiv	ated	-		-		
😻 [DIALOG]Service Activation - Activate										
User Group	dalla	s	-	E>	tension Numb	er	2173			-
Service Type	Pres	et Call Forward Unreachal	ble 💌		Destination		921467432	264		
No Answer Time(sec)	10			ι	Jse Notification	I.				-
Start Time					End Time					
Service Date				A	llow Other Rin	g				-
Preset Call Forward Type	Both		-	Aut	to Record Mailt	ox				
Auto Record Call Type			-	Hot De	sk Expire Time	(hour)				
Hot Line Delay(sec)				Inco	oming Call Log	ging				-
Service Schedule			-							
		A	ctivate Apply	/ _ C	lose					

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]

± ¥			Search
ervice Permission			
Call Forward All	Call Forward Busy	Call Forward No Answer	Call Forward Upreachable
Preset Call Forward All	Preset Call Forward Busy	Preset Call Forward No Answer	Preset Call Forward Unreachable
Preset Call Forward DND	Absence	☑ Wake-Up Call	
🖉 Hot Line	Outbound Call Lock	Auto Retry	☑ ⊂ ···- ☑ Callback
Remote Office	Multi-ring	Caller ID Block	✓ Group Call Forward
No Ring	AME	 ☑ Call Waiting	✓ Auto Answer
Auto Record	✓ Hot Desking	✓ Individual Speed Dial	✓ Incoming Call Logging
Second Service Class	V Follow Me	Caller ID Display	Call Recording
🛿 Add-On Conference	🗹 One-Step Conference	🗹 Music on Hold	🗹 Call Park Extension
🛿 Call Park Orbit	🗹 Call Transfer	🗹 Barge-In with Tone	🗹 Barge-In without Tone
🛛 DND Override	🗹 Restricted Call Forward	Temporary CID Restriction	🗹 Station Paging
🛛 Ring Plan Override	🗹 Paging On Answer	🗹 Remote Extension Set	🗹 Hotel Inter-Room Call Lock
Preminum CID Service	🗹 No Ring Override	🗹 Meet-me Conference	🗹 Multi-Device Conference
🛿 BLF Key Create	🗹 Operator Call	🗹 Call Transfer without Restriction P	🗹 Direct Trunk Selection
🗹 Call Bridge	🗹 Move To Mobile	🗹 Mobile Auto Answer	☑ NFC Service
Smart Routing Activate	Smart Routing including Access Co	Smart Routing Call Reject	🗹 Service After Smart Routing
🛿 Receiving Call in Hot Spot Zone	Receiving Call in mVoIP Zone	🗹 Unregistred BLF On	🗹 NFC Service for Any Mobile
NFC Move to WE VoIP First	Malicious Call Trace		
			Selected Al

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]

[DIALOG]Information - Change			
±			Search
Service Permission			
Call Forward All	🗹 Call Forward Busy	🗹 Call Forward No Answer	🗹 Call Forward Unreachable
🗹 Preset Call Forward All	🗹 Preset Call Forward Busy	🗹 Preset Call Forward No Answer	🗹 Preset Call Forward Unreachable
🗹 Preset Call Forward DND	🗹 Absence	🗹 Wake-Up Call	🗹 DND
🗹 Hot Line	🗹 Outbound Call Lock	🗹 Auto Retry	🗹 Callback
🗹 Remote Office	🗹 Multi-ring	🗹 Caller ID Block	🗹 Group Call Forward
🗹 No Ring	🗹 AME	🗹 Call Waiting	🗹 Auto Answer
🗹 Auto Record	🗹 Hot Desking	🗹 Individual Speed Dial	🗹 Incoming Call Logging
🗹 Second Service Class	🗹 Follow Me	🗹 Caller ID Display	🗹 Call Recording
🗹 Add-On Conference	🗹 One-Step Conference	🗹 Music on Hold	🗹 Call Park Extension
🗹 Call Park Orbit	🗹 Call Transfer	🗹 Barge-In with Tone	🗹 Barge-In without Tone
🗹 DND Override	🗹 Restricted Call Forward	🗹 Temporary CID Restriction	✓ Station Paging
🗹 Ring Plan Override	🗹 Paging On Answer	🗹 Remote Extension Set	🗹 Hotel Inter-Room Call Lock
Preminum CID Service	🗹 No Ring Override	🗹 Meet-me Conference	🗹 Multi-Device Conference
🗹 BLF Key Create	🗹 Operator Call	🗹 Call Transfer without Restriction P	🗹 Direct Trunk Selection
🗹 Call Bridge	🗹 Move To Mobile	🗹 Mobile Auto Answer	☑ NFC Service
Smart Routing Activate	Smart Routing including Access Co	Smart Routing Call Reject	🗹 Service After Smart Routing
Receiving Call in Hot Spot Zone	Receiving Call in mVoIP Zone	☑ Unregistred BLF On	☑ NFC Service for Any Mobile
☑ NFC Move to WE VoIP First	Malicious Call Trace		
			Selected Al
	Change /	Apply Close	

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. Gert the SSID of the wireless LAN the SCMis 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.

Ý 🖻 🐘 🗡 🕞 🔅	MA 9:39 🛐 🔝 🕸
< 🔯 Wi-Fi	
Wi-Fi networks	Scanning
SNAE2G_10 Connected	
ap_5g_ht40 ^{Open}	((1
ARTWORK-1x Secured	
gamma0 ^{Open}	(¢
gamma5 Secured	() F
GENERAL_AP_TEAM Secured	
iptime Secured (WPS available)	
iptime5G	
Scan	Wi-Fi Direct

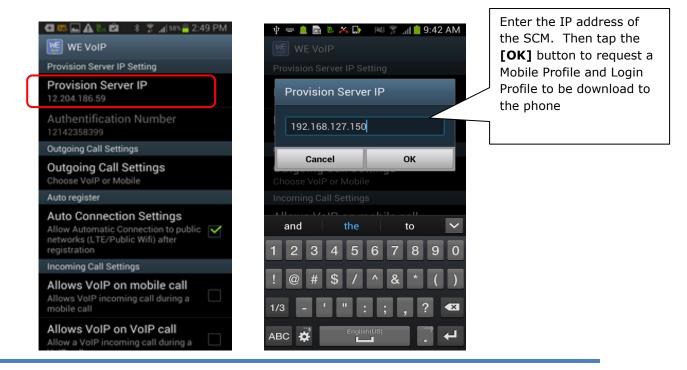
Step 2

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 ${\ensuremath{\$}}$ appears at the top of the screen.

Drag top notification bar down to see the registration status.

If the registration fails, the failure icon $\{ n \}$ appears and the reason for the failure is displayed in a pop up window.



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 settings required for using WE VoIP.

to access the outgoing and incoming call

🐌 🏚 🖄 🛛 🔹 🏋 📶 79% 🧰 9:25 PM	🔜 🖻 🌰 🛍 👘 🕴 🍸 📶 🕬 💼 9:25 PM
🔟 WE VolP	WE VolP
Provision Server IP Setting	Provision Server IP Setting
Provision Server IP 12.204.186.59	Provision Server IP 12.204.186.59
Authentification Number	Authentification Number
Outgoing Call Settings	Outgoing Call Settings
Outgoing Call Settings Choose VoIP or Mobile	Outgoing Call Settings Choose VolP or Mobile
Auto register	Auto register
Auto Connection Settings Allow Automatic Connection to public networks (LTE/Public Wifi) after registration	Auto Connection Settings Allow Automatic Connection to public networks (LTE/Public Wifi) after resistration
Incoming Call Settings	📋 🛱 Request Profile
Allows VoIP on mobile call Allows an incoming VoIP call during a M mobile call if checked	Remote Dial setting Version
Allows VoIP on VoIP call Allows an incoming VoIP call during a	ن version O Exit
	These options appear when you press the MENU button on the smartphone while the WE-VoIP application is open.

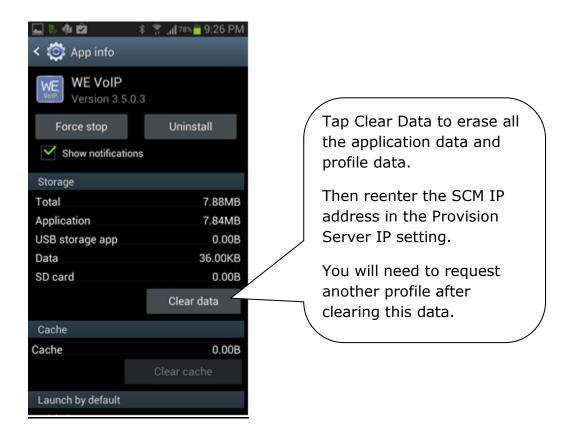
The following table explains the available settings in sequence as you scroll through them.

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	You can choose whether to use VoIP/4G or use 4G only for outgoing calls Choose VoIP or Mobile: You will be prompted to select VoIP or 4G Use only Mobile: All outgoing calls are made over 4G network. Image: Note of the select vole o		
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	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.		
	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.		
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.



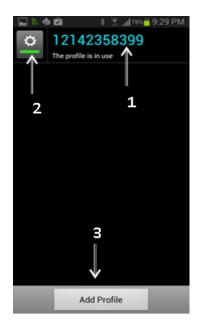
6.3 Administrator Settings

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:

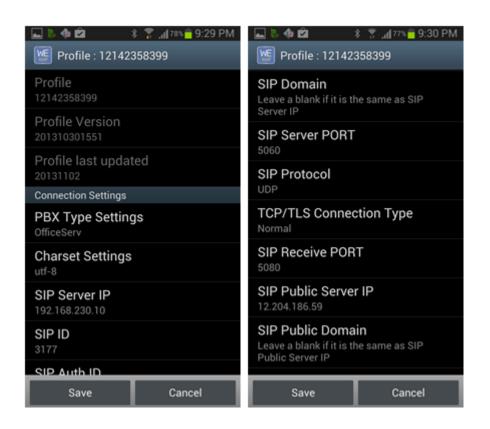
- 1. Click on the phone dialer to use the Dial Pad.
- 2. Dial [1234##**] Do not share this code with users.
- 3. Press the green call button



No.	Name	Function	
0	Profiles	Name and usage status of the WE VoIP profile are displayed. Select this menu to open the edit profile screen. <u>You may have multiple profiles because your connect to other systems (branch office)</u>	
2	Register profile button	Connect to other systems (branch once) 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 <u>Not supported on OfficeServ. Leave as Disabled</u> 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	 You can configure prefix and Digit Map settings for making an external call. 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	 You can configure Wi-Fi related settings. 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	 You can set codec and sound properties to use for a WE VoIP call. Codec Priority: Set audio codecs to use in a WE VoIP call and their priorities. Sound Properties 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	You can set the DTMF method during a WE VoIP call. For OfficeServ always use rfc2833	
Security Settings	You can set the security function available during a WE VoIP call. - 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	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.

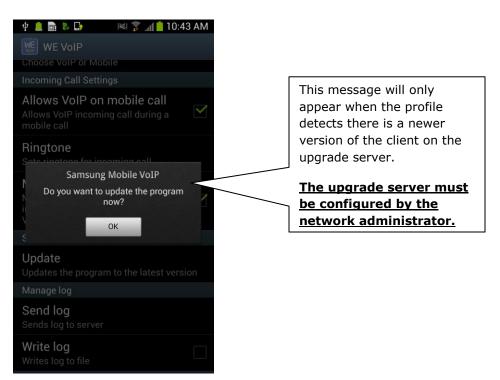
- #!: A prefix is not required for a number starting with #.

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



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.

🔜 🗞 🏟 🖄 👘 🕺 🦹 👔 📶 66% 🧰 10:45 PM	🔜 🗞 🏟 🛍 🛛 🚯 📅 🚮 69% 📅 10:21 PM	ار 🕞 👘 🏟 🖄 👘 🖇 😤 📶 ال 💈 👘
随 WE VoIP	< 🧮 My Files 👘 🚹	< 🚞 My Files 🛛 🔒 🗈
Software update	/storage/sdcard0/smv	/storage/sdcard0/smv
Update Updates the program to the latest version	SMV20131103.log	SMV20131103.log
Manage log	Sip20131103.log	Sip20131103.log
Send log Sends log to server	SMV20131102.log	Delete
Write log Writes log to file	Sip20131102.log	Share via
WE Work Settings	SAE201311020.log	Move
Premium CID settings Sets Premium CID	SAE201311010.log	Сору
Show Context CID		Rename
Disable	SMV20131101.log	Details
Wi-Fi Connection during mVoIP call	C	
mVoIP setting mVoIP settings.	Sip20131101.log	Sip20131101.log
Recording List		
Call recording List		

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