WE VoIP Service

For **Office**Serv™ 7000 Series

WE-VoIP Client V3.5.0.3

OfficeServ V4.65 or higher

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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 Samsung OfficeServ 7000 systems over an industry standard Wi-Fi network.

This mobile SIP client application makes your smart phone a mobile extension of the OfficeServ 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 SIP Stack 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

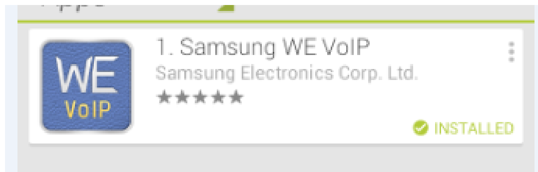
Client Type	Model	Android Version
Samsung Galaxy Edition	Galaxy S3 Series	4.0.3 or higher
	Galaxy S4 Series	4.1 or higher
	Galaxy Note2	4.0.3 or higher

Continue to next page.

Download from Google Play Store

Access Play Store and search 'Samsung WE VoIP'.

This application will appear in the list. Select it.



Then tap 'Install' button to begin the 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 OfficeServ system 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 features are available to WE-VoIP extensions registered to the OfficeServ 7000 system.

Limitations: 1. Service is only available on **SIP** and **PRI** trunks
2. OfficeServ main system software must be **V4.65 or higher.**

	Feature	7030	7100	7200S	7200	7400
1	Connect as a mobile SIP extension	Yes	Yes	Yes	Yes	Yes
2	Can be paired or stand-alone extension	Yes	Yes	Yes	Yes	Yes
3	Make & receive internal VoIP calls over Wi-Fi	Yes	Yes	Yes	Yes	Yes
4	Make & receive outside VoIP calls over Wi-Fi	Yes	Yes	Yes	Yes	Yes
5	Use default dialer of smart phone	Yes	Yes	Yes	Yes	Yes
6	Use Contact list of smart phone	Yes	Yes	Yes	Yes	Yes
7	VoIP calls share Call Logs of smart phone	Yes	Yes	Yes	Yes	Yes
8	Call Hold / Resume	Yes	Yes	Yes	Yes	Yes
9	Call Transfer (Blind transfer only)	Yes	Yes	Yes	Yes	Yes
10	Directed Call pickup of ringing extension	Yes	Yes	Yes	Yes	Yes
11	Group Pickup	Yes	Yes	Yes	Yes	Yes
12	Group Pick Up & Hold Pick Up (12 + XXXX)	Yes	Yes	Yes	Yes	Yes
13	Make station group calls	Yes	Yes	Yes	Yes	Yes
14	Set and Cancel DND using OfficeServ feature code	Yes	Yes	Yes	Yes	Yes
15	VM Message indication (mail icon upper line)	Yes	Yes	Yes	Yes	Yes
16	Missed Call Indication (Not for group calls)	Yes	Yes	Yes	Yes	Yes
17	Move VoIP call from Desk phone to mobile	Yes	Yes	Yes	Yes	Yes
18	Move VoIP call from mobile to desk phone	Yes	Yes	Yes	Yes	Yes
19	Forward when unregistered to Mobile number	Yes	Yes	Yes	Yes	Yes
20	Single CID number for paired WE VoIP ext.	Yes	Yes	Yes	Yes	Yes
21	Register to private IP address of the system	Yes	Yes	Yes	Yes	Yes
22	Register to public IP address of the system	Yes	Yes	Yes	Yes	Yes
23	WE-VoIP over LTE service	Yes	Yes	Yes	Yes	Yes
24	Manual Handover to cellular network "To Mobile"	Yes	Yes	Yes	Yes	Yes
25	Smart Routing changes Mobile call to WE VoIP call	Yes	Yes	Yes	Yes	Yes
26	Auto Updating of Client software by Google Play Store	Yes	Yes	Yes	Yes	Yes
27	Log gathering to assist technical support	Yes	Yes	Yes	Yes	Yes

Hardware	7030	7100	7200S	7200	7400
No New Hardware required for WE VoIP service	**No**	**No**	**No**	**No**	**No**

5. PROGRAMMING PROCEDURES FOR WE VOIP SERVICE

This chapter lists programming procedures in OfficeServ V4.65 of higher, required to set up WE-VoIP service. Each procedure is broken down sections corresponding to the traditional OfficeServ 7000 Series Technical Manual sections:

- General Description
    - This section will describe the purpose of this procedure.
- Programming
    - This section will detail any relevant Device Manager Menu 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 OfficeServ 7000 series 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 OfficeServ 7200S system. When programming a different model system they may appear slightly different. For example cabinets with virtual card slots are not the same for every OfficeServ system. However the procedure is the same.

TIP:

Samsung engineering recommends following these procedures in this order as the most efficient method. If not, you may find some fields will not show in specific DM menus because they are dependent on some other procedure being performed

5.1 Enter WE VoIP License, DM 2.1.4

GENERAL DESCRIPTION

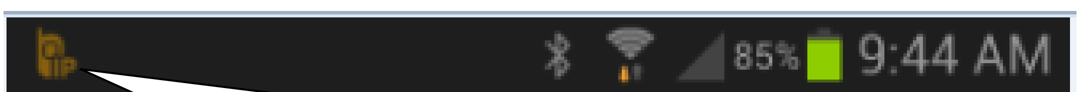
Sites that want to add or start out with WE VoIP extensions must order them as part of the SIP Stack License. Even though WE VoIP users are SIP extensions they are ***not*** included in the SIP Phone max count. They are a separate WE VoIP count.

Example: 20 SIP Phones and 10 WE VoIP are separate license counts, but in **DM 2.7.2 SIP Phone Information**, both SIP phones and WE VoIP phones will be listed.

WE VoIP user license is for concurrent users. So when 20 WE VoIP users are setup in the OfficeServ system, and WE VoIP max count is 10, only 10 WE VoIP users can be registered simultaneously.

When the 11 WE VoIP Client registers and has 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.

Green icon indicates this client has a license and is ready to use

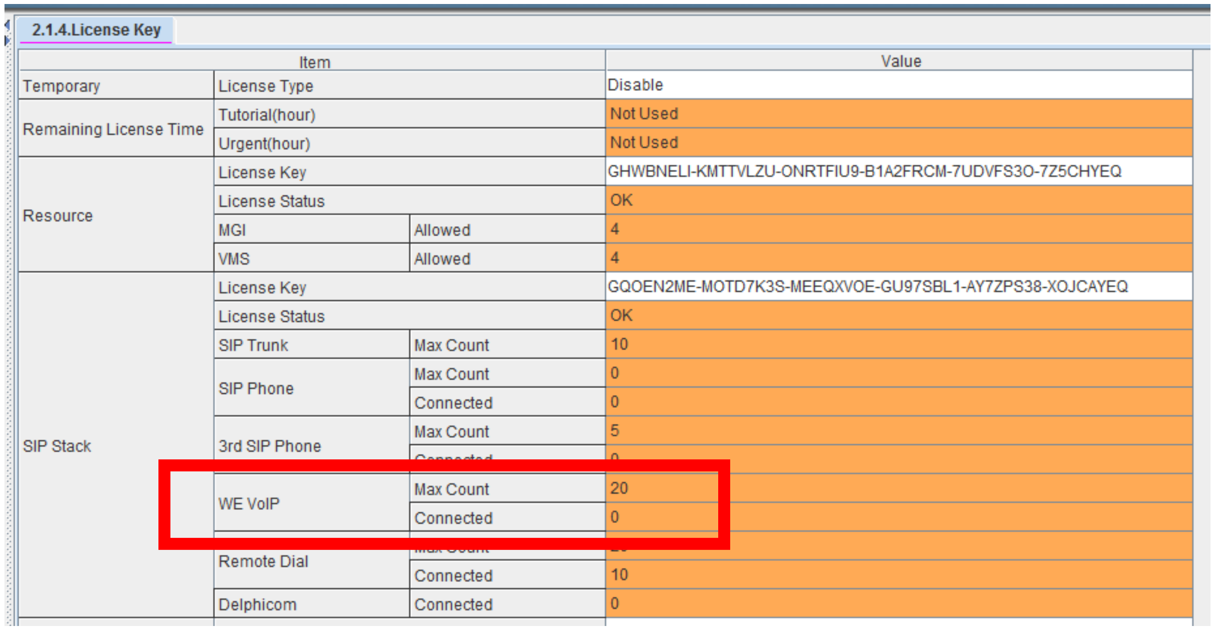


Yellow icon indicates this client does not have a license. It is registered but not ready to use.

PROGRAMMING

WE VoIP Users in SIP Stack License

Upon receipt of the email with the SIP License, copy and paste it into the system using Device Manager Menu, **2.1.4 License Key.** Refresh the screen, and then confirm the max count of WE VoIP users is correct.



2.1.4.License Key

Item			Value
Temporary	License Type		Disable
Remaining License Time	Tutorial(hour)		Not Used
	Urgent(hour)		Not Used
Resource	License Key		GHWBNELI-KMTTVLZU-ONRTFIU9-B1A2FRCM-7UDVFS3O-7Z5CHYEQ
	License Status		OK
	MGI	Allowed	4
	VMS	Allowed	4
SIP Stack	License Key		GQOEN2ME-MOTD7K3S-MEEQXVOE-GU97SBL1-AY7ZPS38-XOJCAYEQ
	License Status		OK
	SIP Trunk	Max Count	10
	SIP Phone	Max Count	0
		Connected	0
	3rd SIP Phone	Max Count	5
		[illegible]	[illegible]
	WE VoIP	Max Count	20
		Connected	0
	Remote Dial	[illegible]	[illegible]
		Connected	10
	Delphicom	Connected	0

When needed, check the WE VoIP "connected" field to see how many of the users are registered (connected).

Note: The 60 day Tutorial License supports the maximum WE VoIP count for 60 days.

OfficeServ System	7030	7100	7200-S	7200	7400
MAX WE-VoIP Clients	16	32	56	56	224

5.2 Assign Virtual Card Slots, DM 6.3.2

GENERAL DESCRIPTION

WE VoIP service requires one or more virtual card slots to be assigned for the following;

1. **SIP Stations**

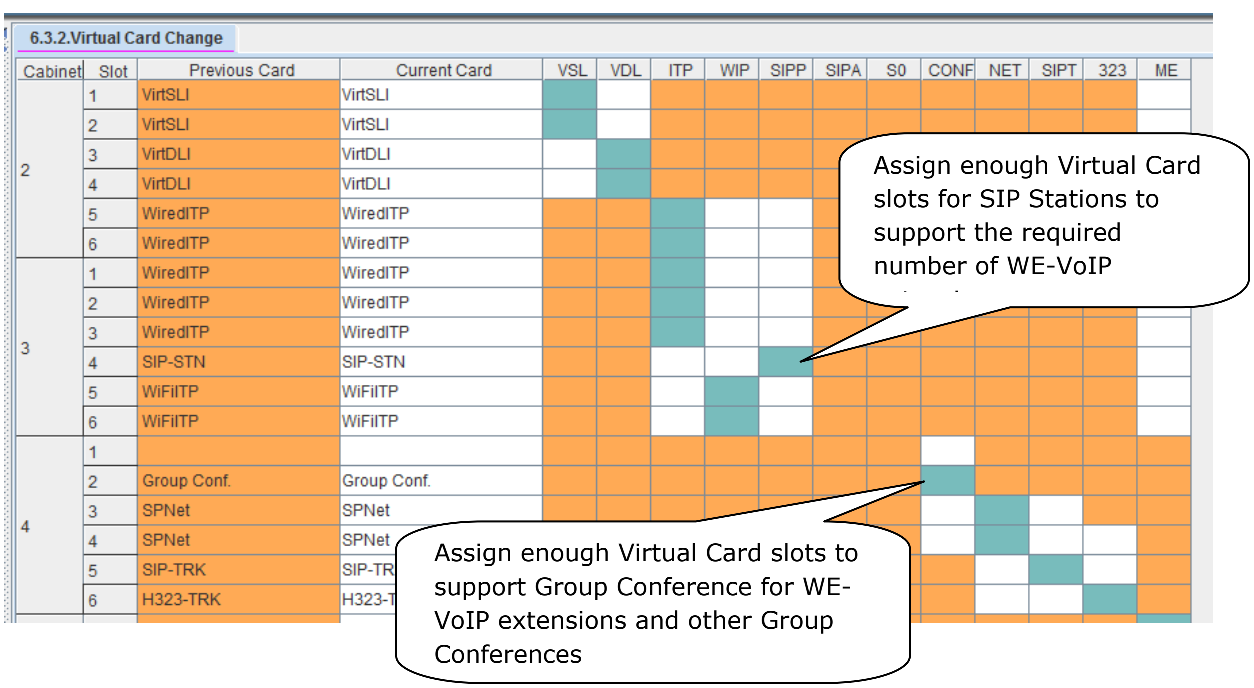
   WE VoIP Service requires at least one Virtual Card Slot be assigned for SIP Stations (SIP-STN). Assign enough to meet the WE VoIP max count plus other SIP stations as needed.

2. **Conference Groups**

   The “Move to Mobile” & “Move to Desk” feature requires Group Conference feature. Assign enough Virtual Card Slots to meet WE VoIP max count plus other Group Conferences as needed. This is not a one –to – one match for WE-VoIP clients as it is unlikely all of them will use the Move feature at the same time.

PROGRAMMING

Device Manager Menu **6.3.2 Virtual Card Change** is used to assign SIP-STN slots and Group Conf. slots as required.



5.3 Update Numbering Plan, DM 2.8.0

GENERAL DESCRIPTION

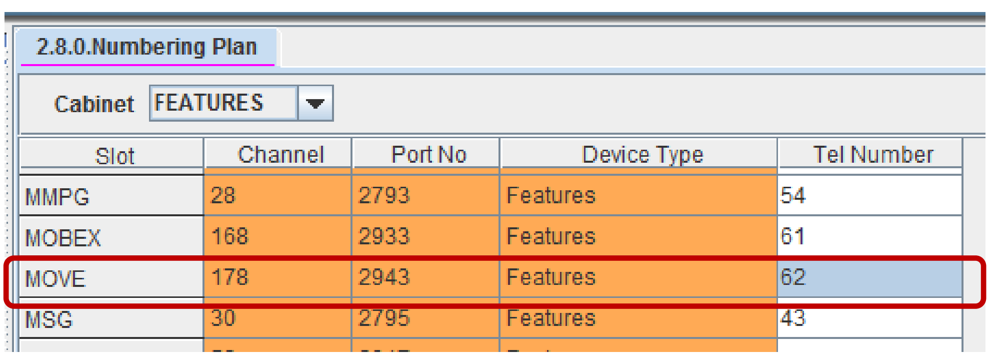
The system numbering plan should be modified to provide for the following;

1. Assign numbers to SIP stations for WE VoIP users. This **SIP-STN** number is the WE-VoIP user's extension. Assign one for each WE-VoIP Client. These numbers will appear in DM 2.7.2 SIP Phone Information.
2. Assign numbers to Mobex Stations (**MOBEX STN**). This Mobex Station number is used to call the mobile cell phone number when the WE-VoIP Client is unreachable. This is covered in section 5.11 of this document. These Mobex numbers will appear in DM 2.7.5 Mobile Extension.
3. Assign a feature assess code to the **Move** feature.

PROGRAMMING

Device Manager Menu **2.8.0 Numbering Plan** is used to assign numbers to MOVE Feature Code, SIP Stations and Mobex Stations.

Assign Move Feature Code



Assign SIP Station Numbers

2.8.0.Numbering Plan

Cabinet C3

Slot	Channel	Port No	Device Type	Tel Number
	8	255	IP Phone	3232
S3	1	256	IP Phone	3233
	2	257	IP Phone	3234
	3	258	IP Phone	3235
	4	259	IP Phone	3236
	5	260	IP Phone	3237
	6	261	IP Phone	3238
	7	262	IP Phone	3239
	8	263	IP Phone	3240
S4	1	264	SIP Phone	2172
	2	265	SIP Phone	2173
	3	266	SIP Phone	2174
	4	267	SIP Phone	2175
	5	268	SIP Phone	2190
	6	269	SIP Phone	2177
	7	270	SIP Phone	2185
	8	271	SIP Phone	
	1	272	WIP Phone	3301
	2	273	WIP Phone	3302

This example shows 7 SIP Phones assigned extensions numbers that will be used for each WE VoIP extension (Client)

Assign MOBEX station numbers

2.8.0.Numbering Plan

Cabinet C6

Slot	Channel	Port No	Device Type	Tel Number
S1	1	384	MOBEX STN	8801
	2	385	MOBEX STN	
	3	386	MOBEX STN	8802
	4	387	MOBEX STN	
	5	388	MOBEX STN	8803
	6	389	MOBEX STN	
	7	390	MOBEX STN	8804
	8	391	MOBEX STN	
S2	1	392	MOBEX STN	8805
	2	393	MOBEX STN	
	3	394	MOBEX STN	8806
	4	395	MOBEX STN	
	5	396	MOBEX STN	8807
	6	397	MOBEX STN	
	7	398	MOBEX STN	8808
	8	399	MOBEX STN	
S3	1	400	MOBEX STN	8809
	2	401	MOBEX STN	
	3	402	MOBEX STN	8810
	4	403	MOBEX STN	

This example shows 10 MOBEX extensions assigned numbers 8801 to 8810.

Reminder: Only assign numbers to odd numbered ports. MOBEX feature uses them in pairs.

5.4 Network Configuration

GENERAL DESCRIPTION

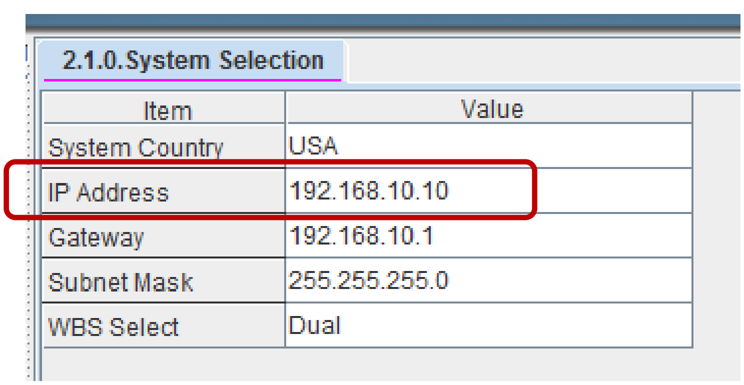
The following network configuration settings must be completed to enable WE-VoIP service. The settings are in several different Device Manager Menus. Because we are assuming this is a working system many of these setting have been completed. Some only need to be verified and or modified as detailed below.

The system IP addresses and will need to be provided to the Client users. They need to enter these during the Client registration process on the individual smart phones.

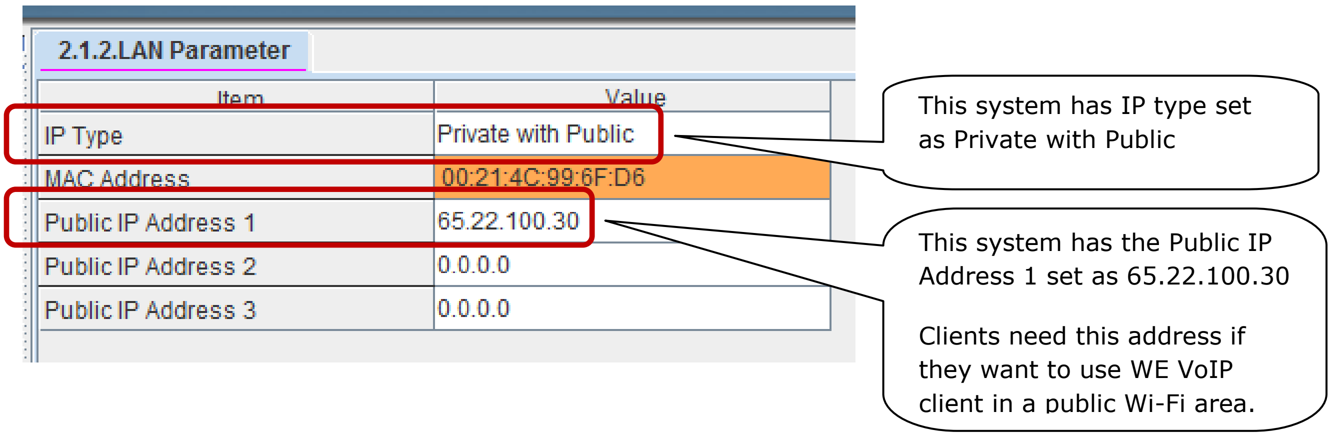
1. Enter or confirm the system's **Private IP address**. Clients will need this to register to the phone system. *Your working system will already have this assigned*.
2. Enter or confirm the system's **Public IP address**. To use Wi-Fi in a public area (Not office Wi-Fi network) Client will need to register to the system's public IP address. *Your working system may or may not already have this assigned.*
3. Set the system's **IP Type** to **Private with Public.** *Your working system may or may not already have this assigned.*
4. Set MGI Card (MGI16/64/OAS) **IP Type** to **Private with Public.** *Your working system most likely will already have this assigned.*

PROGRAMMING

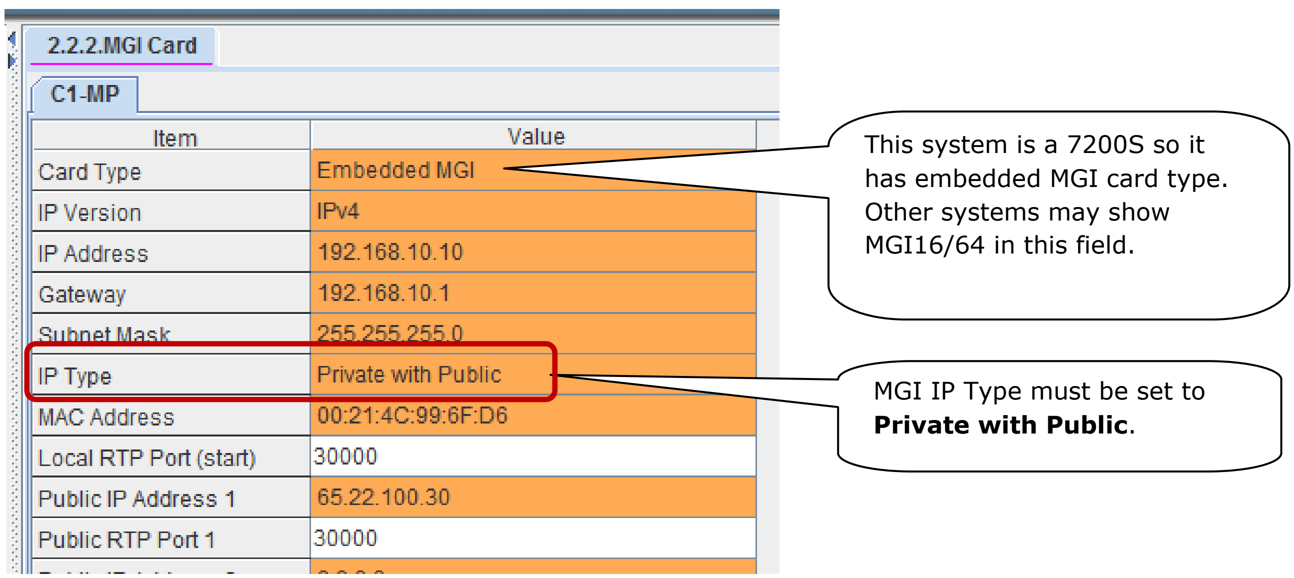
Device Manager Menu **2.1.0. System Selection** is used to assign the system IP address information**.**



Device Manager Menu **2.1.2. LAN Parameter** is used to assign the system **IP Type** and Public IP Address.



Device Manager Menu **2.2.2. MGI Card** is used to assign the **MGI IP Type**



5.5 Define Mobile Extensions, DM 2.7.5

GENERAL DESCRIPTION

WE VoIP Service only uses Mobile Extensions (MOBEX) when the WE VoIP Client is unreachable. When it is unreachable (disconnected), incoming calls to the client will be forwarded to the Mobile Extension which is a pre-assigned public telephone number of the client's smart phone.

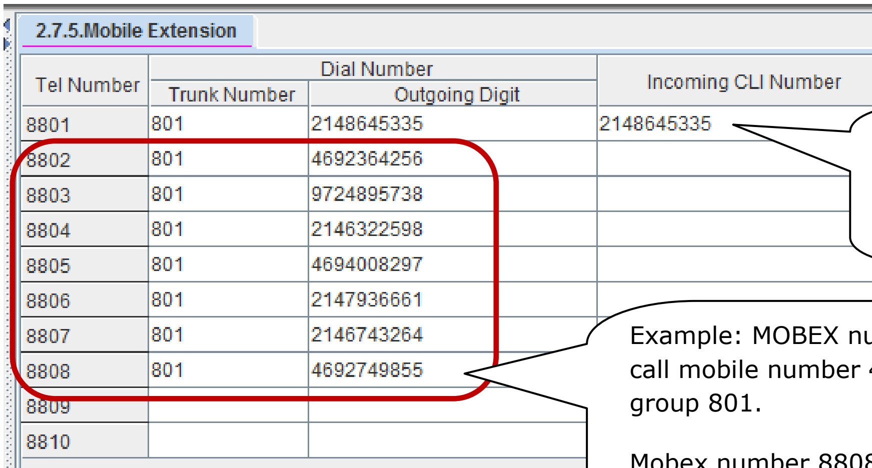
WE-VoIP extensions can work without a Mobile Extension, but will be limited in capability when they are disconnected or unreachable.

In DM 2.7.5 Mobile Extension, assign the Trunk Number and Outgoing Digits for each MOBEX number that will be associated with a WE-VoIP client. MOBEX numbers are assigned in DM 2.8.0 Numbering Plan as detailed in section 5.3.

1. **Trunk Number**: Enter the trunk or trunk group number to be used for outgoing calls. Must be a PRI trunk group or SIP trunk group. **Required for WE VoIP Service**
2. **Outgoing Digits**: This is the 10 digit telephone number of the WE-VoIP Client's smart phone. This is the pre-assigned mobile telephone number the system dials to call the Clients smart phone in the Call Forward Unreachable condition. **Required for WE VoIP Service**
3. **Incoming CLI Number:** This is the incoming CLI received when this phone calls into Executive Mobex. It is used to associate and authorize this Mobex user to access the system. ***Not needed for WE VoIP Service***.
4. **CallBack:** Used for Executive Mobex callback service**.** ***Not needed for WE VoIP Service.***
5. **AA Go to Voice Mail**: Set to "Yes" to enable Voice Mail message notification to Mobex Executive station. ***Not needed for WE VoIP Service.***
6. **Master Station**: Enter the corresponding SIP extension to the WE VoIP client. This is used to match the mobile extension to the SIP client when they are unreachable. This is the same SIP extension that is entered in the **Call Forward Unreachable** field in DM 2.7.2 SIP Phone Information.
7. **Status**: Used for Executive Mobex ports if they will take calls or not. **Not needed for WE VoIP Service.**

PROGRAMMING

Device Manager Menu **2.7.5 Mobile Extensions** is used to assign the telephone number of the mobile smart phone that will be used as the WE VoIP Client.

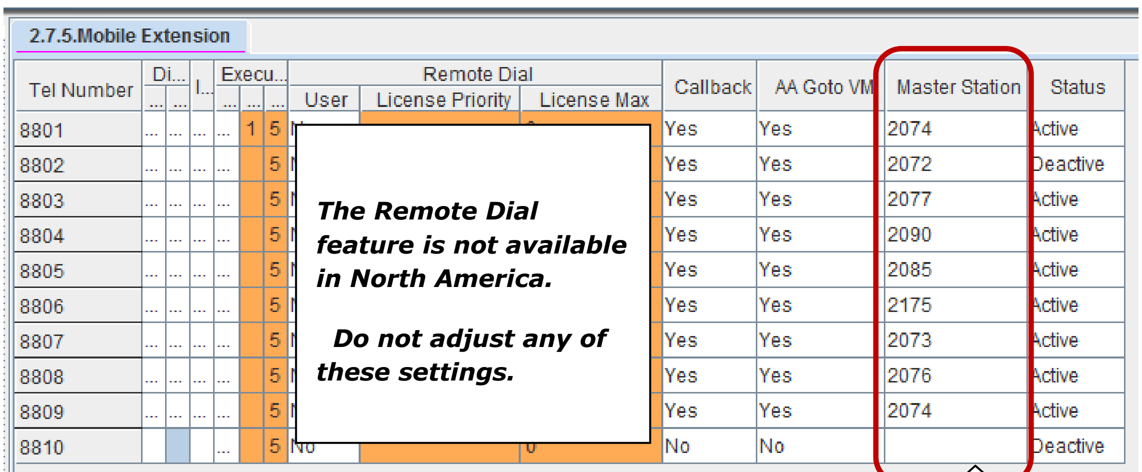


Tel Number	Dial Number: Trunk Number	Dial Number: Outgoing Digit	Incoming CLI Number
8801	801	2148645335	2148645335
8802	801	4692364256	
8803	801	9724895738	
8804	801	2146322598	
8805	801	4694008297	
8806	801	2147936661	
8807	801	2146743264	
8808	801	4692749855	
8809			
8810			

This line shows 8801 is an Executive Mobex number and is not related to WE-VoIP Service.

Example: MOBEX number 8808 is assigned to call mobile number 469-274-9855 using trunk group 801.

Mobex number 8808 is the Call forward unreachable destination assigned in DM 2.7.2 SIP Phone Information.



Tel Number	Di...		I...	Execu...			Remote Dial: User	Remote Dial: License Priority	Remote Dial: License Max	Callback	AA Goto VM	Master Station	Status
	...	...		...	...	...							
8801	...	...	...	...	1	5				Yes	Yes	2074	Active
8802	...	...	...	...		5				Yes	Yes	2072	Deactive
8803	...	...	...	...		5				Yes	Yes	2077	Active
8804	...	...	...	...		5				Yes	Yes	2090	Active
8805	...	...	...	...		5				Yes	Yes	2085	Active
8806	...	...	...	...		5				Yes	Yes	2175	Active
8807	...	...	...	...		5				Yes	Yes	2073	Active
8808	...	...	...	...		5				Yes	Yes	2076	Active
8809	...	...	...	...		5				Yes	Yes	2074	Active
8810				...		5	No		0	No	No		Deactive

Enter SIP Client extensions that correspond to the mobile extension.

5.6 SIP Phone Information, DM 2.7.2

GENERAL DESCRIPTION

Each WE-VoIP Client is assigned in the OfficeServ system as a SIP Phone. The SIP phone numbers assigned in DM 2.8.0 Numbering Plan, appear in DM 2.7.2.

Use DM 2.7.2 Sip Phone Information to assign the follow:

1. **User ID**: Enter a User ID for each SIP Phone number. *With the new security measures in V4.65 or higher the SIP User ID is blank.*
    a. This can be the same as the extension number
    b. Or something like sip_xxxx used in the example below. The sip_xxxx ID format makes it more difficult to register unauthorized SIP phones than simply duplicating the extension number.
2. **Password**: Enter a password. *With the new security measures in V4.65 or higher, the password is blank.* *The new password must be at least 6 digits up to a maximum of 8 digits.*
3. **Call Forward Unreachable**: In this field enter the associated Mobile Extension used in DM 2.7.5. The system will forward the incoming call to this number under two unreachable conditions;
    a. **Case 1: Plug Out** – 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.
    b. **Case 2: No Response** – In case the WE-VoIP client does not respond to an invite message (180/183 ringing) from the OfficeServ system within the value set in the Call Forward Unreachable Time, the incoming call to the client will be forwarded to the pre-assigned number in the Call Forward Unreachable field. This time is set in DM 5.14.2
4. **Insert Trunk Port:** Insert a trunk number or the trunk group access code that is to be used for the WE-VoIP extensions. Must be SIP or PRI. It must be the same trunk group that is used in DM 5.2.23 Mobile Profile, Auto Prefix Code. Generally it will be the dial "9" group.
5. **Use IP White List:** *With the new security measures in V4.65 or higher, the IP White list is blank.* It is recommended that this be set to "**Disable**" for each WE-VoIP client extension. If not, you must enter the IP address of each smart phone. The IP address may change each time the client registers to the OfficeServ. It is not practical to use IP White list for WE-VoIP SIP extensions.

PROGRAMMING

Device Manager Menu **2.7.2 SIP Phone Information**

Tel Number 3405 is a standard desk SIP phone, not a WE VoIP SIP Client.

2.7.2.SIP Phone Information

Tel Number	User ID	Password	Tone Source	Call Waiting	Call Foward Unreachable
3403	5555	******	Use System Tone	Enable	
2173	sip_2173	******	Use System Tone	Disable	8807
2190	sip_2190	******	Use System Tone	Disable	8804
2185	sip_2185	******	Use System Tone	Disable	8805
2174	sip_2174	******	Use System Tone	Disable	8801
2177	sip_2177	******	Use System Tone	Disable	8803
2176	sip_2176	******	Use System Tone	Disable	8808
2175	sip_2175	******	Use System Tone	Disable	8806
2172	sip_2172	******	Use System Tone	Disable	8802
2178	sip_2178	******	Use System Tone	Disable	8809

These are SIP stations used for WE-VoIP Client extensions.

These are the Mobile Extensions assigned in DM 2.7.5 that are used when the WE VoIP SIP Client is unreachable.

2.7.2.SIP Phone Information

Tel Number	User ID	...	...	...	...	...	Insert Trunk Port	Insert Trunk Type	NAT Check	Use IP White List
3403	5555	...	...	...		...	7001	Disable	IP	Enable
2173	sip_2173	...	...	...	...	...	801	Disable	IP	Disable
2190	sip_2190	...	...	...	...	...	801	Disable	IP	Disable
2185	sip_2185	...	...	...	...	...	801	Disable	IP	Disable
2174	sip_2174	...	...	...	...	...	801	Disable	IP	Disable
2177	sip_2177	...	...	...	...	...	801	Disable	IP	Disable
2176	sip_2176	...	...	...	...	...	801	Disable	IP	Disable
2175	sip_2175	...	...	...	...	...	801	Disable	IP	Disable
2172	sip_2172	...	...	...	...	...	801	Disable	IP	Disable
2178	sip_2178	...	...	...	...	...	801	Disable	IP	Disable

Enter the LCR access code, TRK number or TRK Group to be inserted. When dialing numbers from the SIP phone call log this access code is automatically inserted. PRI or SIP trunks must be in this trunk group for WE-VoIP Service.

5.7 Create Mobile Profile, DM 5.2.23

GENERAL DESCRIPTION

All WE VoIP Clients use the system Mobile Profile. There is only one. The first time the WE-VoIP Client application is started it will register to the OfficeServ system and download the Mobile Profile. Each time the application is started then stopped it will check for the latest Mobile Profile that matches the IP address of the OfficeServ system it is registering to.

> ***Notice***
> ***Any changes made to the Mobile Profile will require each WE VoIP Client to request a profile update from their smart phone.***

The Mobile Profile sets and controls the following attributes that all WE-VoIP Clients use.

1. **AP SSID:** This identifies the local Wi-Fi network that the OfficeServ system will be connected to. Obtain this from the Network administrator. The WE-VoIP Client will connect to this wireless network before it can register to the IP address of the OfficeServ system.

2. **Upgrade Server:** Needed for WE-VoIP software upgrade. If available, enter the IP address of the server that will have the WE-VoIP client software installed. Clients will use this to update when a new version of the Samsung WE VoIP software is made available. Initially it will need to be manually loaded form a PC to each smart phone via a USB. See section 3 of this document.

3. **Upgrade URL:** If using the upgrade server, enter the URL here.

4. **Auto Prefix Code:** This is the trunk access code that will automatically be prefixed to a dialed number. Generally it is **"9"** if only one trunk group is available or LCR is enabled. If not using the same trunk group as desk phones, then enter the trunk group access code for the SIP or PRI trunks to be used by the WE-VoIP clients. The access code must be the one that corresponds to the trunk group used in DM 2.7.2 SIP Phones Information / Insert Trunk Port field. This Prefix code will be displayed in front of numbers dialed using the WE-VoIP selection.

5. **Auto Prefix Exception List:** The auto prefix code will not be prefixed when dialing numbers in the exceptions list. When dialing any of the sample exceptions listed below, the auto prefix code will not be inserted in front of these.

XXX	Any 3 digits. 3 digit extensions, feature codes like 401-DND set & 400-DND Cancel. Or 55+1 for page zone 1
XXXX	Any 4 digits, 4 digit extensions

65!	65 + any dialed number
*64	Any dialed number plus 64
911	Special numbers
XXXXX	Any 5 digits >3 digit extensions plus 2 digit feature codes Example: 12+201 for hold pickup, 65+201 for direct pick up
XXXXXX	Any 6 digits 4 digit extensions plus 2 digit feature codes like 12+2001 for hold pickup, 65+2001 for direct pick up

Note: **X**s used in Prefix Exception List must be all CAPS

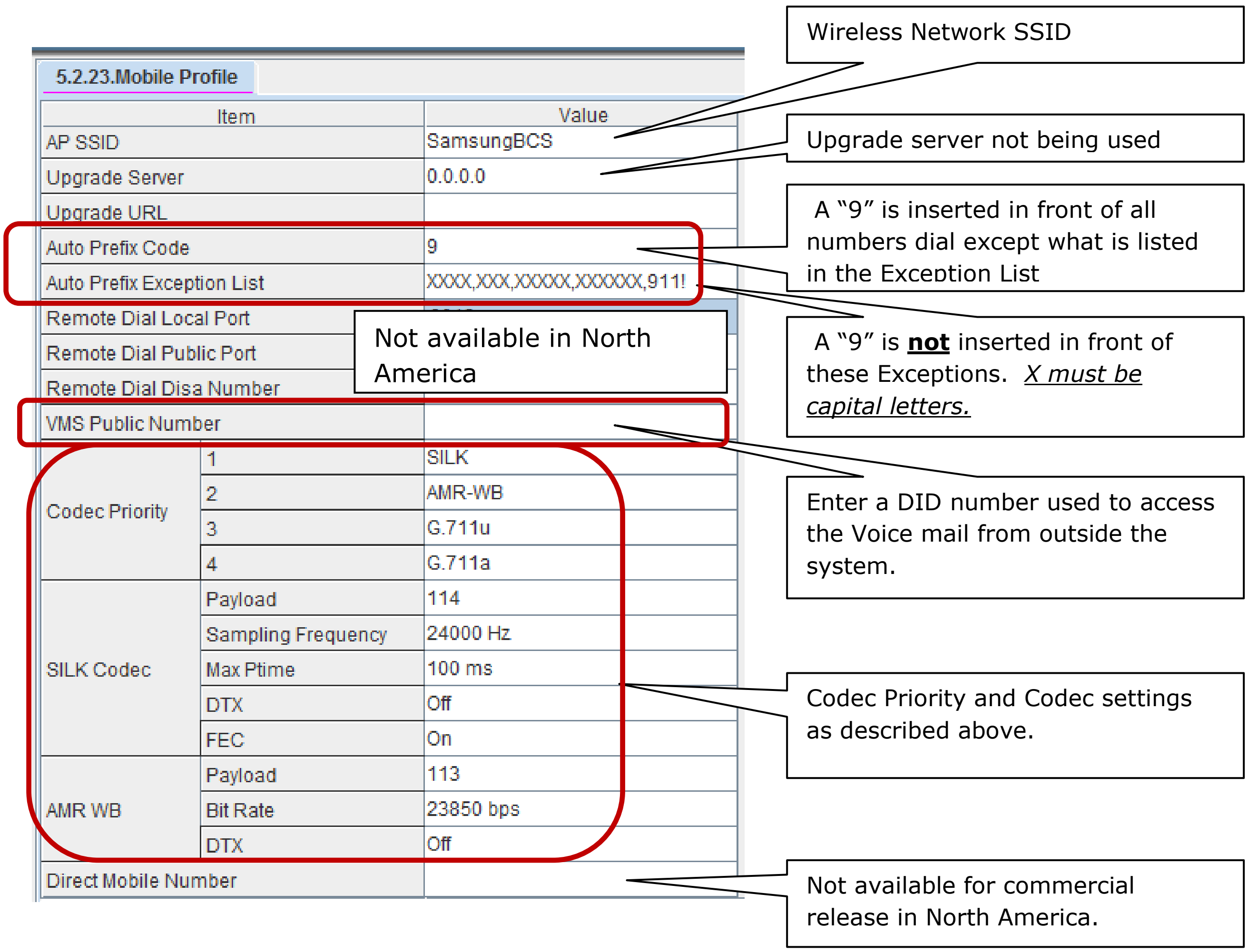
6. **Remote Dial Local Port:** Not available in North America. Do not change.
7. **Remote Dial Public Port:** Not available in North America. Do not change.
8. **Remote Dial DISA Number:** Not available in North America. Do not change.
9. **VMS Public Number:** This is the DID number assigned to access the VM from outside the system.
10. **Codec Priority:** The Client will use the SILK codec as first choice then move through the four priorities as set. Each priority number can select any of the codec choices. For now leave these as default.
11. **Silk Codec:** Set SILK codec options.
    a. Payload
    b. Sampling Frequency
    c. Max Ptime – Max packet time
    d. DTX – Select whether DTX (Discontinuous TX) is used or not. When DTX is enabled, packet is not sent in case of silent. So network load is decreased. (We recommend you set this option as OFF for voice quality stability.)
    e. FEC – Select whether FEC (Forward Error Correction) is used or not. When FEC is enabled, the phone has noise robust feature. (We recommend you set this option as ON for voice quality improvement.)

12. **AMR WB:** Set AMR-WB codec options.
    a. Payload
    b. Bit Rate
    c. DTX – Select whether DTX (discontinuous TX) is used or not. When DTX is enabled, packet is not sent in case of silent. So network load is decreased. (We recommend you set this option as OFF for voice quality stability.)

13. **Direct Mobile Number:** When dialing this number, system makes an outgoing call to 4G network. Currently not used in North America.

PROGRAMMING

Device Manager Menu **5.2.23 Mobile Profile** settings



5.8 Create Login Profile, DM 5.2.24

GENERAL DESCRIPTION

Each client must have a Login Profile that assigns their Mobile telephone to their WE-VoIP extension. This is the most important part of the Login Profile. You must input the Mobile number for each extension. All other default settings are optional. Profiles identify the specific attributes of each WE VoIP client. Assign the following for each:

1. **Mobile Number:** Assign the mobile telephone number to the WE-VoIP extension created in SIP numbering plan. This identifies who is WE VoIP extension 2xxx. This number will be used to create a unique file name for both the Mobile and Login profile like <sec_mobile_19724895738.xml>, <sec_login_19724895738.xml>
    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.
2. **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.)
3. **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.)
4. **AECM**: Select Auto Echo Cancellation Mode. (Default: Speaker Phone). Unless you are provided some special instruction, do not change default value.
5. **Echo:** Select whether Echo Cancellation is used or not. (Default: Enable). Unless you are provided some special instruction, do not change default value.
6. **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.
7. **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

8. **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.
9. **DTMF Type:** Select DTMF type. Use RFC2833 for We-VoIP
10. **Public Zone Service:** To use WE VoIP Service in a public area (Public Wi-Fi or LTE) this option should be set to "Enable". *If not the phone will not register outside the Office Wi-Fi area.*
11. **Scan 5G Only:** To reduce handover delay, select the channel scanning option that the Access Points can meet. **Default is 5G Only**
    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.
12. **Use sRTP:** Enable or Disable secure RTP option as required by the Network Administrator. (Default: Disable)
13. **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.
14. **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.
15. **TOS:** Set IP header TOS field for RTP media. Adjust TOS bits as required by the Network Administrator. (Default: 11100000)
16. **Jitter:** 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.
17. **SIP Signal Type:** Select the signaling packet type UDP, TCP or TLS. (Default: UDP). TLS is recommended when user will frequently be using public Wi-Fi. See section 5.17 Firewall Settings.
18. **Description:** This field can be used to enter a brief 16 character description of the WE VoIP Client for easy reference. Example: Eddie Galaxy S3 (Default: blank).

PROGRAMMING

Device Manager Menu **5.2.24 Login Profile** must be completed for each WE-VoIP client extension.

As illustrated below only **Mobile Number**, **Public Zone Service** and **Scan 5G Only** needs to be completed. The remaining data fields can remain as default data unless Network Administrator requires changes.

5.2.24.Login Profile

Tel Number	Mobile Num	No...		...	...	Sw...		...	...	Public Zone Service	Scan 5G Only	USE sRTP
		...	Tx			...	Tx					
3403		...	...	...	...	...	...	...	..	Disable	5G Only	Disable
2173	12146743264	...	...	...	...	...	...	...	..	Enable	Auto	Disable
2190	12146322598	...	...	...	...	...	...	...	..	Enable	2.4G Only	Disable
2185	4694008297	...	...	...	...	...	...	...	...	Enable	Auto	Disable
2174	12148645335	...		...	...	...	...	...	...	Enable	Auto	Disable
2177	19724895738	...				...	...	...	...	Enable	Auto	Disable
2176	14692749855	...	...				...	...	...	Enable	Auto	Disable
2175	12147936661	...	...						...	Enable	Auto	Disable
2172	14692364265	..	...	...						Enable	Auto	Disable
2178		...	...	...						nable	Auto	Disable

Note: The cell phone carrier for 469-400-8297 does not use the "1" but another carrier for WE-VoIP extension 2172 requires the "1"

These are all of the data fields in DM 5.2.24 Login Profile that are detailed above.

5.2.24.Login Profile

Tel Number	Mobile Num	Noise Suppression		Aecm	Echo	Swing Free		Cng	DTMF Type	Public Zone Service
		Rx	Tx			Rx	Tx			
3403		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable
2173	12146743264	Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Enable

Scan 5G Only	USE sRTP	Multi Frame	Multicast	TOS	Jitter	SIP Signal Type	Description
5G Only	Disable	Disable	Disable	11100000	4	UDP	
5G Only	Disable	Disable	Disable	11100000	4	UDP	Eddie Galaxy III
5G Only	Disable	Disable	Disable	11100000	4	UDP	Mike Galaxy III

5.9 Station Pair & Single CID Number Service, DM 4.2.1

GENERAL DESCRIPTION

A WE VoIP Client extension can be **stand alone** or **paired.** When paired with another phone three types of service are provided.

1. Multi-Ring Service

When a user receives an incoming call on a paired desk phone or WE VoIP phone both devices will ring to avoid missed calls. Setting or clearing Call Forward or DND options will be applied to both devices.

2. Single CID Number

When two stations are paired, decide which station will send caller ID for ***internal*** calls. Set Single CID Number to ON for this station and the other is automatically set to OFF. Example: 2072 deskphone is paired with 2172 WE-VoIP phone. Single CID Number is set to ON for 2072. When either of these two phones make an intercom call, the called phone receives CID of 2072.

Use DM 2.4.3 **Send CLI Number** to set single CID number for **external** calls. Set the CLI for both paired extensions to send the same CLI number.

3. Voice Message Notification Control

Basic Voice Mail setup will configure a mailbox for the desk phone extension and a mailbox for the WE VoIP SIP Client extension. They are in fact two separate extensions and may have individual greetings. When paired in DM 4.2.1, setting the **Single CID Number** option to **ON** for one of the paired numbers will also determine that this device will receive VM Message Notification. When replying to VM Message notification by pressing the Message button on desk phone or Message Icon on WE-VoIP Client the user is always logged into the mailbox of the station set to **ON.** Example: 2072 deskphone is paired with 2172 WE-VoIP phone. Single CID Number is set to ON for 2072. An unanswered call to either extension will result in the VM Message notification being delivered to 2072.

Note: Dialing the VM group pilot number will log user into the mailbox of the station they are call from.

PROGRAMMING

DM Menu 4.2.1 **Station Pair** is used to set the option described above.

4.2.1.Station Pair

Primary No	Secondary No	Single CID Number
3403		Off
2173	2073	Off
2190	2090	Off
2185	2085	Off
2174	2074	Off
2177	2077	On
2176	2076	Off
2175	2075	Off
2172	2072	Off
2178		Off
2071		Off
2072	2172	On
2073	2173	On
2074	2174	On
2075	2175	On
2076	2176	On
2077	2177	Off
2078		Off

Desk phone 2072 is paired with 2172

Single CID Number is set to **On** for 2072 and **Off** for 2172

2072 will send the CID and it will receive VM Message notifications.

5.10 Setup Call Move Feature

GENERAL DESCRIPTION

WE VoIP Service provides a very convenient way to move a VoIP call between a paired desk phones and a WE VoIP client with the press of the pre-assigned MOVE button. The VoIP calls move in both directions: WE VoIP Client extension number and Deskphone extension number must be paired in DM 4.2.1.

Deskphone to WE VoIP Client
While on a call, pressing the MOVE button on the deskphone moves the call to the WE VoIP client (Smart Phone) and automatically disconnects the deskphone.

WE VoIP Client to Deskphone
While on a WE VoIP call on the Smart Phone, pressing the MOVE button on the paired deskphone moves the call to the deskphone and automatically disconnects the WE VoIP call on the WE-VoIP client.

Restrictions:
Although other types of phones can be paired with the WE VoIP client there are two restrictions.

1. In case of SLI and WIP phones, Call Move is only supported by dialing the pre-assigned Move feature code. DM 2.8.0 Number Plan in section 5.3 of this document.
2. Only WE-VoIP Client can support the Call Move feature. Other SIP phones and SIP clients cannot use Call Move feature.

USER INSTRUCTIONS

Call Move from Deskphone to WE VoIP Client

a. A user is on a call on the deskphone.
b. User presses the 'MOVE' key on the paired deskphone.
c. The paired WE VoIP client (Smart Phone) will ring.
d. User answers the ringing call and is connected to the caller. The deskphone is automatically disconnected.

Call Move from WE VoIP Client to Deskphone

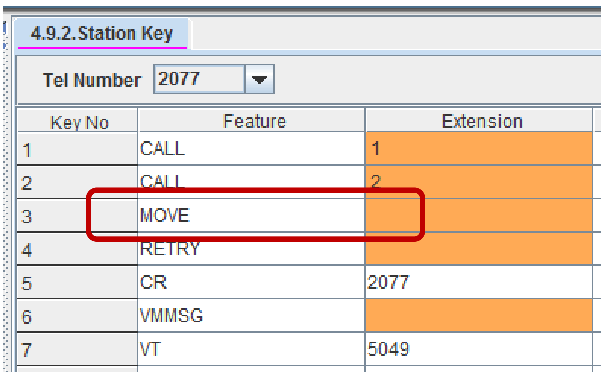
a. A user is on a VoIP call on his WE-VoIP Client (Smart Phone).
b. User returns to his office and presses the pre-assigned MOVE button on the paired deskphone.
c. The WE-VoIP client is disconnected from the call in progress

d. The call is moved to the deskphone with **no ringing**.

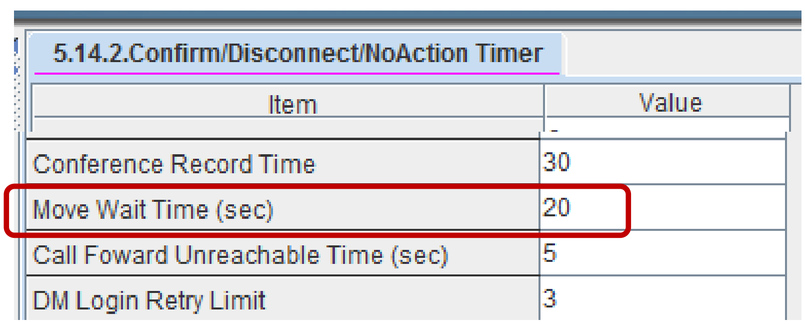
PROGRAMMING

DM Menu 4.2.1 **Station Pair** is used to set the station paring for deskphone and WE VoIP client. This is covered in section 5.0 of this document.

DM 4.2.9 **Station Key** is used to assign the **MOVE** key to any button on the deskphone.



Device Manager Menu **5.14.2 Confirm/Disconnect/NoAction Timer** is used to set the Move Wait Time. **Default is 20 seconds**



Note: When moving a call from deskphone to WE-VoIP client: If the client does not receive a call within the Move Wait Time value, the call to the WE VoIP client is disconnected and the call remains on the deskphone.

5.11 Setup Call Forward Unreachable

GENERAL DESCRIPTION

When the WE VoIP SIP client is unreachable for the reasons listed below, the system will forward the call to the **Call Forward Unreachable** destination entered in DM 2.7.2 SIP Phone information.

The unreachable destination is the Mobile Extension associated with the WE-VoIP client in DM 2.7.5, already covered in section 5.5 of this document.

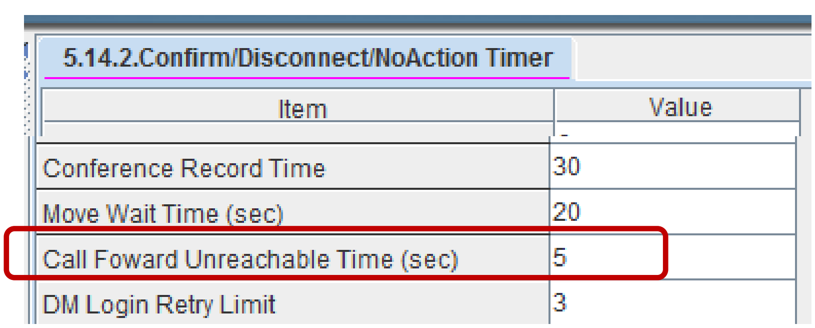
a. **Case 1: Plug Out** – When a WE-Voip client is disconnected normally (unregistered), an incoming call to the client will be forwarded to the pre-assigned number in the Call Forward Unreachable field.

b. **Case 2: No Response** – In case the WE-VoIP client does not respond to an invite message (180/183 ringing) from the OfficeServ system within the value set in the Call Forward Unreachable Time, the incoming call to the client will be forwarded to the pre-assigned number in the Call Forward Unreachable field. This time is set in DM 5.14.2

PROGRAMMING

In DM 2.7.2 **SIP Phone Information**, check to see that the Call **Forward Unreachable** field has the corresponding Mobile Extension number as covered in section 5.6 of this document.

Device Manager Menu **5.14.2 Confirm/Disconnect/NoAction Timer** is used to set the Call Forward Unreachable Time. **Default is 5 seconds**


5.14.2.Confirm/Disconnect/NoAction Timer

Item	Value
Conference Record Time	30
Move Wait Time (sec)	20
Call Foward Unreachable Time (sec)	5
DM Login Retry Limit	3

5.12 Basic System Settings for WE VoIP Service

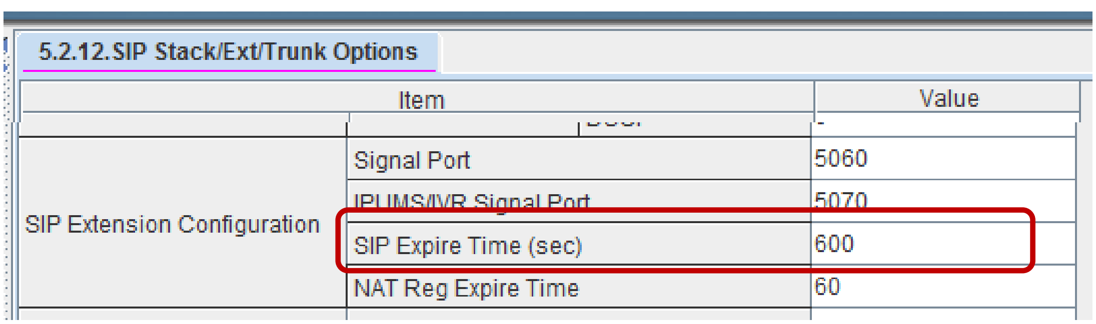
GENERAL DESCRIPTION

There are several system settings that need to be set to support WE-VoIP Service.

1. DM 5.2.12 **SIP Stack/EXT/Trunk Options > SIP Extension Configuration** Set SIP Expire Time to 600 seconds or below. It cannot exceed 600 seconds. If it does exceed 600 seconds mobile client does not try to register.
2. **CODEC Limitation**: Engineering recommends changing codec of IPP/SIP/WIP/SIP Trunk/SPNet to G.711 when using WE-VoIP Service because WE-VoIP phone does not support G.729. So when codec of IPP/SIP/WIP/SIP Trunk/SPNet is G.729 there will be voice problems because of mismatching codec.

PROGRAMMING

Device Manager Menu **5.2.12 SIP Stack/EXT/Trunk Options > SIP Extension Configuration**



5.13 WE VoIP Outgoing Digits

GENERAL DESCRIPTION

Use this menu when it is necessary to modify the outgoing digits form a WE-VoIP client

PROGRAMMING

Device Manager Menu **5.2.29**

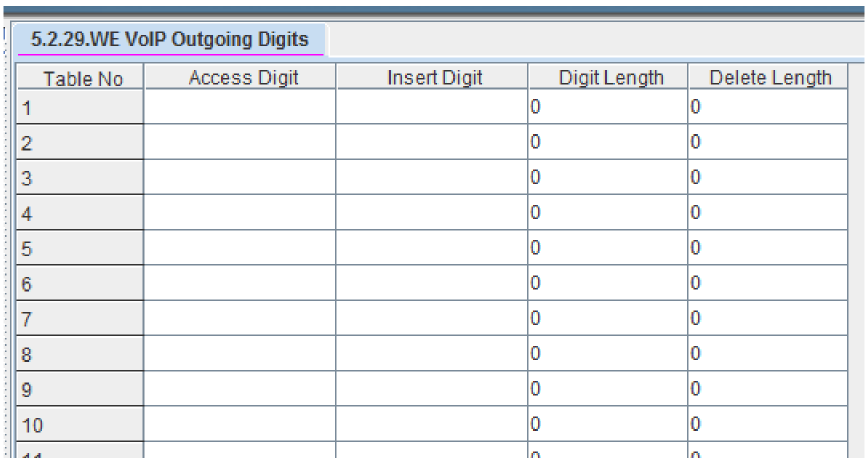


Table No	Access Digit	Insert Digit	Digit Length	Delete Length
1			0	0
2			0	0
3			0	0
4			0	0
5			0	0
6			0	0
7			0	0
8			0	0
9			0	0
10			0	0

5.14 WE VoIP Incoming Caller ID Modify

GENERAL DESCRIPTION

Use this menu when it is necessary to modify the incoming Caller ID/CLI digits to a WE-VoIP client.

PROGRAMMING

Device Manager Menu, **5.2.30 WE-VoIP Incoming Caller Modify** is used to modify the incoming caller ID number.

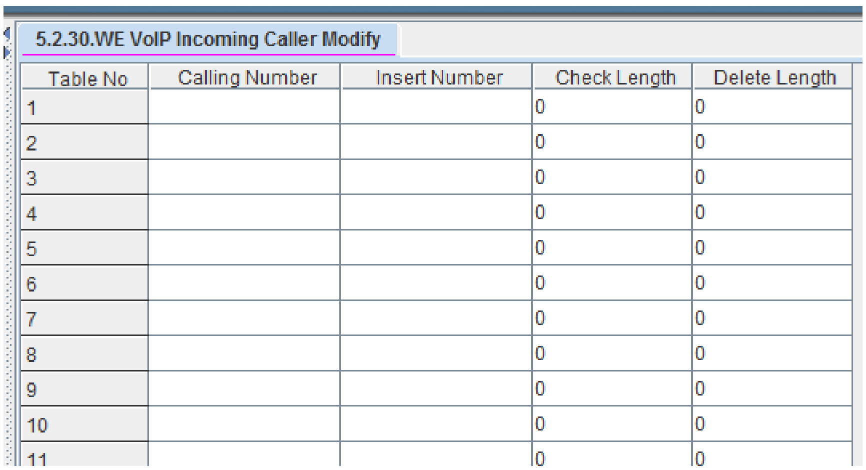

5.2.30.WE VoIP Incoming Caller Modify

Table No	Calling Number	Insert Number	Check Length	Delete Length
1			0	0
2			0	0
3			0	0
4			0	0
5			0	0
6			0	0
7			0	0
8			0	0
9			0	0
10			0	0
11			0	0

5.15 WE VoIP Roaming Channels

GENERAL DESCRIPTION

The Network administrator may decide to use only selected specific channels to manage handoff and roaming on the Wi-Fi network.

PROGRAMMING

Device Manager Menu **5.2.31 WE VoIP Common Options** is used to select which will be scanned for roaming for the 2.4G and 5G frequencies

5.2.31.WE VoIP Common Option

Item		Value
Roaming Scan 2.4G	Channel 1	Off
	Channel 2	Off
	Channel 3	Off
	Channel 4	Off
	Channel 5	Off
	Channel 6	Off
	Channel 7	Off
	Channel 8	Off
	Channel 9	Off
	Channel 10	Off
	Channel 11	Off
	Channel 12	Off
	Channel 13	Off
Roaming Scan 5G	Channel 36	Off
	Channel 40	Off
	Channel 44	Off
	Channel 48	Off
	Channel 149	Off
	Channel 153	Off
	Channel 157	Off
	Channel 161	Off
	Channel 165	Off

Set each channel **On** or **Off** as required by the WLAN Administrator.

Default: OFF

all channels are set to not to be scanned for roaming.

5.16 Smart Phone Model Parameters

GENERAL DESCRIPTION

Different Smart Phone models may require different setting to manage Roaming. Consult with the Wireless Network Administrator before making any changes.

PROGRAMMING

Device Manager Menu **5.2.27 WE VoIP Model Parameters** is used to modify on a per Smartphone model basis as required.

Only change these settings when you receive special instructions from Samsung Technical Support or Samsung Engineering.

5.2.27.WE VoIP Model Parameter

Entry No	Model Name	Roaming Trigger	Roaming Delta	Roaming Scan Period
1	default	-70	10	3
2	SHW-M250S	-70	10	3
3	SHW-M250K	-70	10	3
4	SHW-M250L	-70	10	3
5	SHV-E120S	-70	10	3
6	SHV-E120K	-70	10	3
7	SHV-E120L	-70	10	3
8	SHV-E160S	-70	10	3
9	SHV-E160K	-70	10	3
10	SHV-E160L	-70	10	3
11	SGH-I777	-70	10	3
12	SGH-i747	-70	10	3

5.17 Firewall Settings

GENERAL DESCRIPTION

To enable the WE-VoIP Client to register to the public IP address of the system when using public Wi-Fi or 4G/LTE open the following ports:

Port **80** using either UDP or TCP. This is the port that the WE VoIP Client uses to get the login profile.

Port **9012** using either UDP or TCP. This is the port the WE VoIP Client uses for control signaling to the system main processor.

Port **5060** for SIP signaling using UDP or TCP

Optional > Port **5061** for SIP signaling using **TLS** over UDP or TCP. This is encouraged when using the WE VoIP Client over public Wi-Fi.

PROGRAMMING

<u>Firewall settings are not part of the OfficeServ System. Consult the local LAN administrator to make these setting in the firewall.</u>

When the WE VoIP Client is used over a public Wi-Fi connection and you want to use TLS go to DM 5.2.24 Login Profile and set SIP Signaling Type to TLS as shown below.

5.2.24.Login Profile

Tel Number		DTMF Type	Public Zone Service	Scan 5G Only	USE sRTP	Multi Frame	Multicast	TOS	Jitter	SIP Signal Type
3403		RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000	4	UDP
2173		RFC2833	Enable	5G Only	Disable	Disable	Disable	11100000	4	TLS
2190		RFC2833	Enable	5G Only	Disable	Disable	Disable	11100000	4	UDP

5.18 LCR Suggestion

GENERAL DESCRIPTION

Contacts imported from Outlook will generally have telephone numbers programmed as 1+10 digits. These contacts were created like this because the PSTN needs 11 digit numbers. The OfficeServ LCR will strip off the 1 for local area codes. Numbers added to the list of phone contacts are 10 digits because the cellular network does not need a 1.

So using a contact in your phone with stored with only 10 digits may not go through when selecting the WE VoIP option in phone dialer. If you PRI trunks go to the PSTN the LD number needs a "1". Your SIP provider may or may not require the "1". Cell phone networks never need the "1" but will ignore it if dialed.

So to accommodate these mixed environments it is recommended that the LCR tables be programmed to dial the number with or without the "1".

PROGRAMMING

In Device Manager Menu **3.1.2** Routing Digits makes the necessary entries to meet your site dialing requirements. For example:

Entry No.	LCR Digit	Length	Route Table
1	1	11	1
2	2	10	2
3	3	10	2
4	4	10	2
5	5	10	2
6	6	10	2
7	7	10	2
8	8	10	2
9	9	10	2

Route Table 1 to use Modify Digits entry 1 >insert a "9" and send all 11 digits.

Route Table 2 to use Modify Digits entry 2 > insert "91" in front of a 10 digit number.

Local area codes will have to be broken into specific entries for each local prefixes in that area code.

6. Using the WE VoIP Client

6.1 Registering the WE VoIP to Client OfficeServ

Registering the WE-VoIP Client to the Provisioning Server is the process for registering to the IP address of the OfficeServ System. This can be the Private or Public IP address as determined by the network administrator and company policy. If you do not use the public IP address the WE VoIP client cannot register to the system when out of the office.

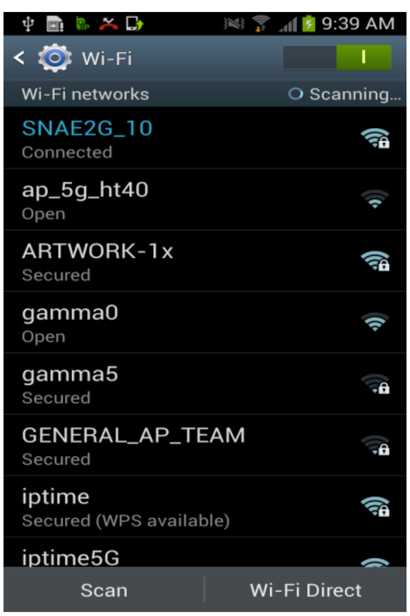
CHECKLIST

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 OfficeServ system is connected to.
3. Get the WE VoIP provision server information. This is the IP address of the OfficeServ 7000 system.

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.*

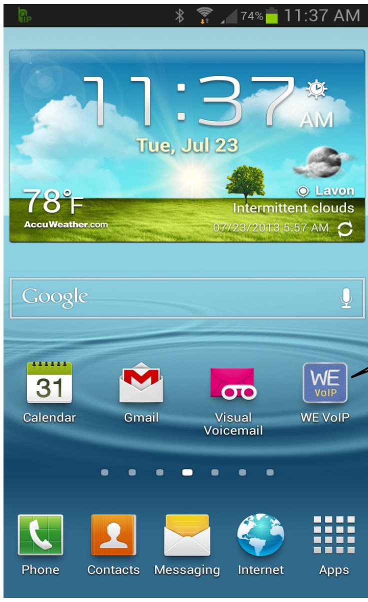
Step 1

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

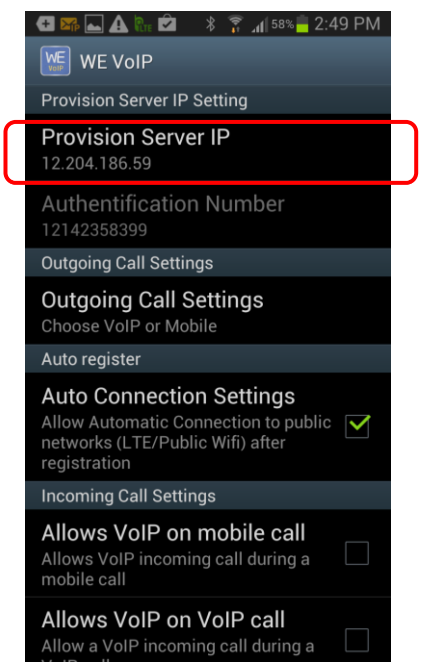
The WE VoIP Client application should already be installed on the device as detailed in section 3 of this manual. Open the WE-VoIP application by tapping on the WE-VoIP Icon.

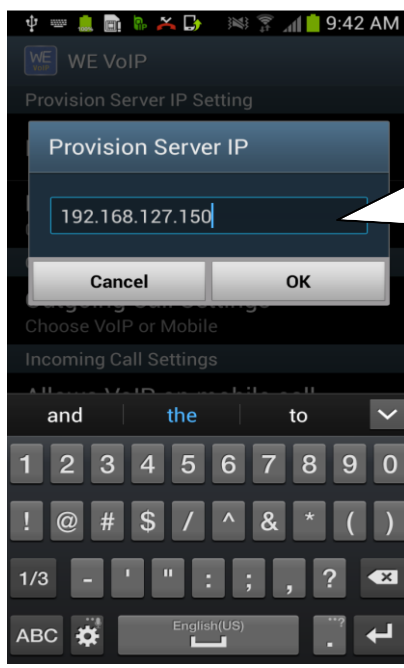


Application Icon added to the home screen of a Galaxy S3

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.



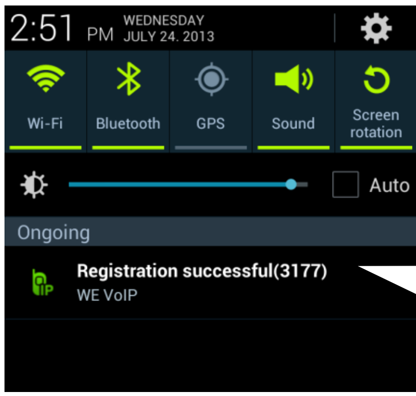


Enter the IP address of the OfficeServ 7000 system. Then tap the **[OK]** button to request a Mobile Profile and Login Profile to be download to the phone

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 idicates 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

NOTE: *You must request a new profile every time the Mobile Profile in the OfficeServ system is changed or every time the provisioning IP address is changed*

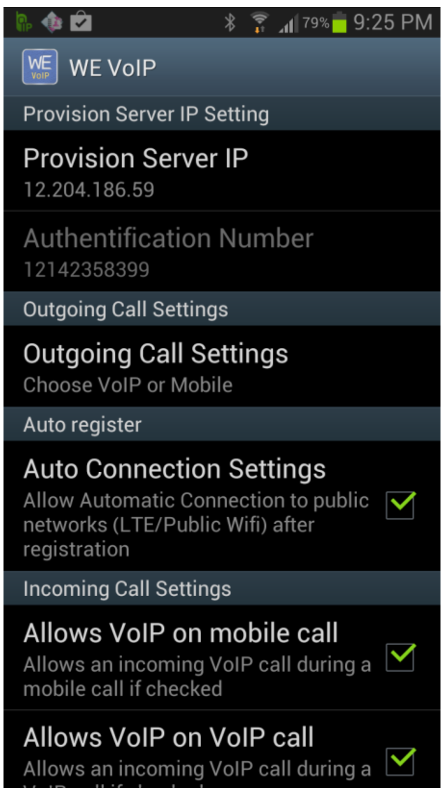
6.2 Client Main Menu Settings

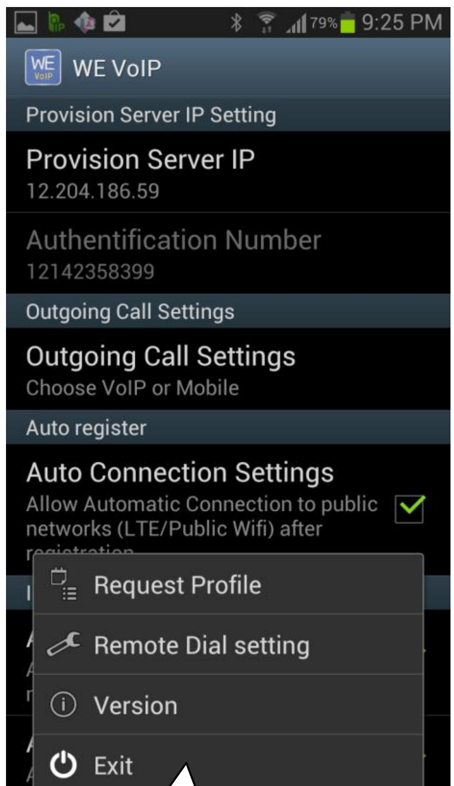
USER SETTINGS

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.

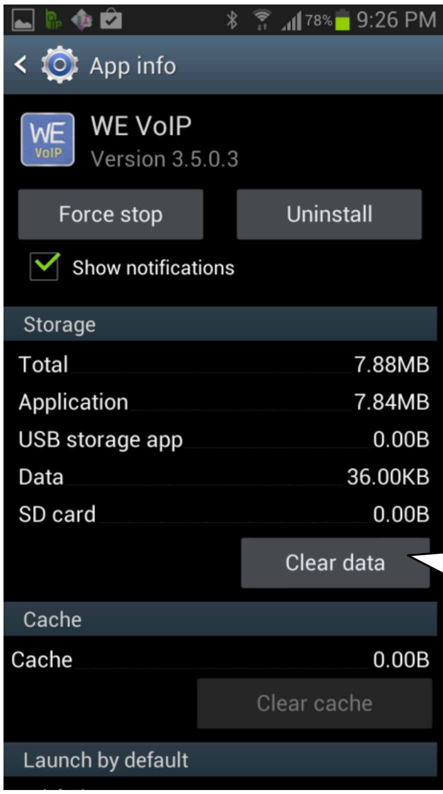
The following table explains the available settings in sequence as you scroll through them.

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	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. NOTE: Even if Choose VoIP or Mobile is checked, outgoing calls are made over 4G network if you are not logged into WE VoIP.
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. NOTE: 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

Menu	Description
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.
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 OfficeServ IP address in the Provision Server IP setting.

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

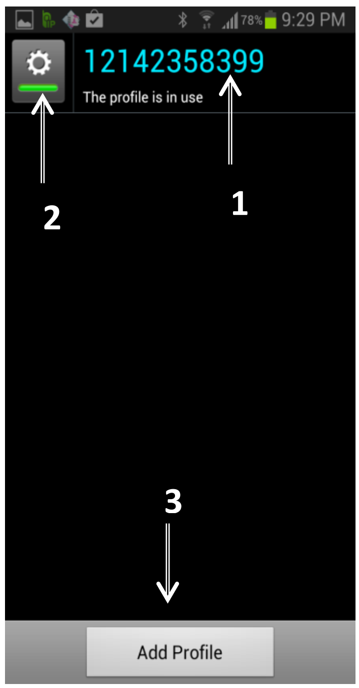
6.3 Administrator Settings

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:

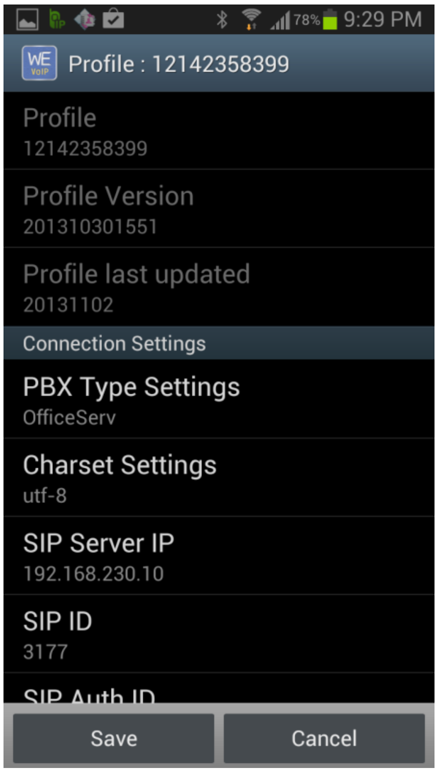
1. Click on the phone dialer to use the Keypad.
2. Dial [**1234##****] *Do not share this code with users.*
3. Press the green call button. This screen will appear.

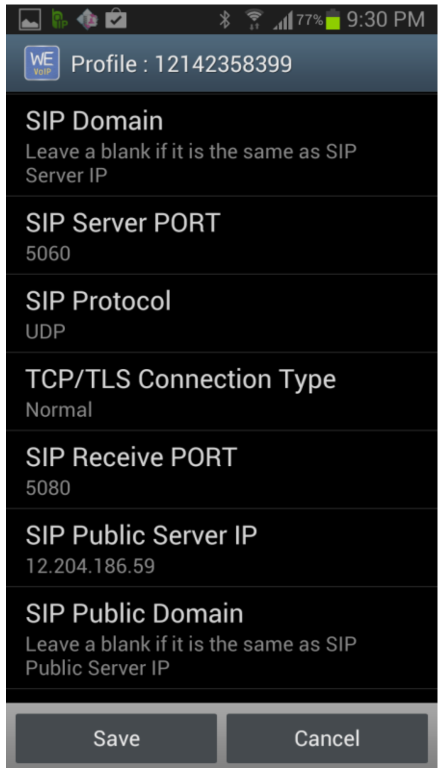


	Name	Function
1	Profiles	Name and usage status of the WE VoIP profile are displayed. Select this menu to open the edit profile screen. *You may have multiple profiles because your connect to other systems (branch office)*
2	Register profile button	This button registers the profile with the company's 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 screens.

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.

When finished 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
Every setting related to the profile is 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 *Not supported on OfficeServ. Leave as Disabled* 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. *Not supported on OfficeServ*
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 on OfficeServ
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. Not supported on OfficeServ
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	Ignore these settings. Not supported on OfficeServ
Menu → Delete Profile	You can delete a profile. If you delete a profile, the profile will also be deleted from the administrator settings screen.

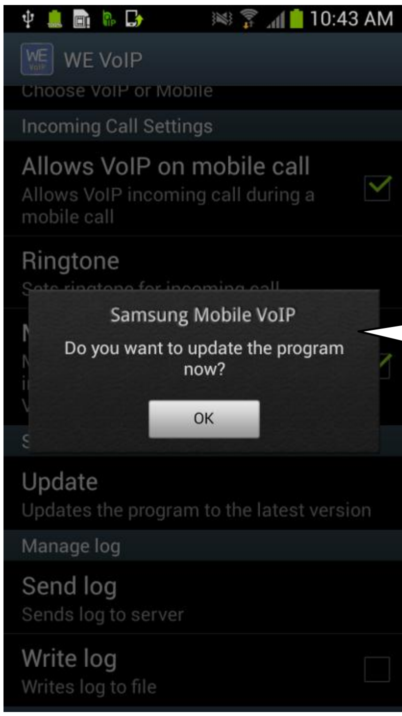
NOTE

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

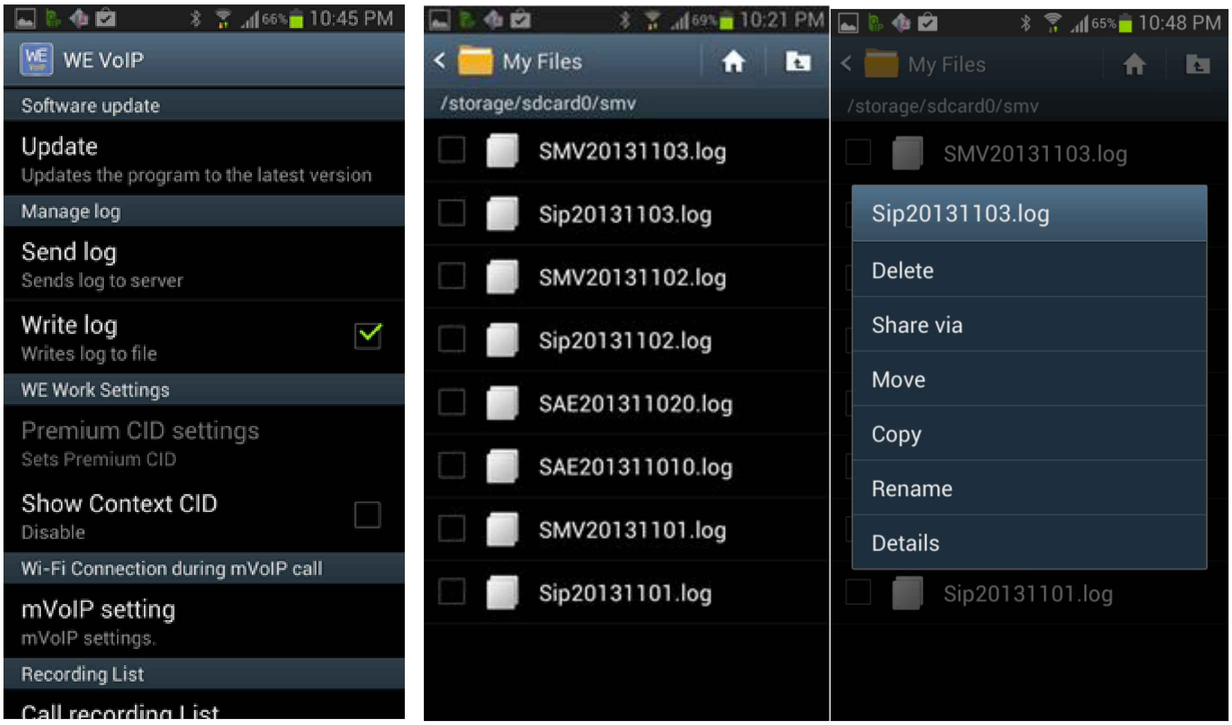


This message will only appear when the profile detects there is a newer version of the client on the upgrade server.

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