

The background of the slide is a dark blue gradient. It features a complex, abstract graphic of a globe or sphere on the left side, composed of a dense grid of small dots and connected by thin lines, creating a wireframe effect. The rest of the background is filled with a network of thin, light blue lines connecting various points, suggesting a global or digital network.

Media5-fone Pro for iOS

Revision 01

2016-01-21

Table of Contents

1 Getting Started	4
1.1 Media5-fone for ios	4
1.2 Media5-FoneThird-Party Software Copyright Information	4
1.3 Starting the Media5-Fone for the First Time	4
1.4 Configuring VoIP Network Service	6
1.5 Configuring Wi-Fi Access Points	6
1.6 Modifying Your Login Password	6
2 Contacts and Contact Lists	7
2.1 Adding a New Contact	7
2.2 Adding a Contact via the Dialler	8
2.3 Changing the Filter of the History Log	8
2.4 Viewing the Communication History Log	8
2.5 Deleting the History Log	8
3 Speed Dialing	10
3.1 Setting Up a Contact for Speed Dialing	10
3.2 Modifying the Number of a Favorite Contact	10
3.3 Removing a Contact from Speed Dialing	10
3.4 Calling a Favorite Contact (Speed Dialing)	11
4 Audio Calls	12
4.1 Calling with the Dialler	12
4.2 Calling a Contact	12
4.3 Ending a Call	12

4.4 Putting a Call on Hold	13
4.5 Putting a Call on Mute	13
4.6 Transferring a Call to Another Contact	13
4.7 Making a Second Call	14
4.8 Merging Calls	14
4.9 Swapping from One Call to Another	14
4.10 Checking Your Voicemails	15

5 Settings 16

5.1 Setting the Language of Media5 Client	16
5.2 Changing the Ringtone	16
5.3 Enabling Notifications	16
5.4 Enabling Call Waiting	16
5.5 Removing a Contact from the Black List	17
5.6 Blocking Contact Numbers	17
5.7 Modifying the Speaker Volume	18
5.8 Purchasing Features	18
5.9 Settings for iOS Devices	18

6 Account Settings 20

6.1 Account Settings	20
6.2 Modifying Account Settings	34
6.3 Creating an Account	34
6.4 Deleting an Account	35
6.5 Listening to Music while Using Media5-Fone	35

7 Copyright Notice 36

1 Getting Started

1.1 Media5-fone for iOS

The Media5-fone is a Client that runs on iOS devices version 7.0 and higher. It is a SIP Client (softphone) that enables users to make and receive VoIP (Voice over Internet Telephony) calls.

VoIP calls are calls established by using a Wi-Fi or 3G/4G mobile packet data network coverage using the IP technology of the Media5-fone. This manual describes how to configure and use the Media5-fone Client. It does not describe how to operate the other functionalities of the iOS device. Please refer to the device's documentation for details on how to operate it.

1.2 Media5-Fone Third-Party Software Copyright Information

The Third Party Software Copyright Information for Media5-fone lists the third-party software modules used in the Media5-fone along with any copyright and license information.

This document is available at: [Third-Party Software Copyright Information](#)

1.3 Starting the Media5-Fone for the First Time

Information

Note: This step is not required if you are upgrading your Media5-fone Client.

When the Client is first started, it automatically detects if it is the first time and displays the Welcome screen.

Steps

- 1) In the Welcome screen, tap **Start**.
- 2) Select how you wish to configure your account
- 3) Tap **Preconfigured List** to select one of the multiple pre-configured IP PBXs and SIP Application Servers accounts or configured Internet Telephony Service Providers accounts.
- 4) Tap your SIP provider.
- 5) Enter your credentials username and password.
- 6) Or,

- 7) Tap **Define Manually** to manually define your account or use a SIP provider not listed.
- 8) Configure the required parameters.

Note: Refer to Creating an Account for details on each and every account parameters that are customizable in this mode.

- 9) Tap **Save**.

Result

Note: The Media5-fone DOES NOT automatically create accounts with any provider. You will need to apply for a service contract with the provider you select to use. Media5 Corporation is not affiliated with any of the providers listed in the configuration wizard. The list is merely a convenience to ease the Client's setup. Providers that wish to be listed here should contact Media5's Technical Assistance Center.

Note: If you already own a SIP account on your company's VoIP server, you do not need any additional SIP provider account. Please contact your IT department to help you configure your SIP account into the Media5-fone.

Note: Media5-fone applies predefined configuration values that are known to work with the selected SIP provider. Should you need to change any settings for your account, you will be able to do so once the configuration wizard is finished.

Note: Media5 Corporation is interested in hearing from you should your provider be unlisted. Media5 would like to get the most complete list of SIP providers around the world. Should you

encounter any issue with a pre-configured provider account or if your provider is not listed, please contact our Technical Assistance Center.

1.4 Configuring VoIP Network Service

Information

You must obtain a VoIP network service configuration prior to using the Media5-fone Client. Please ask your IT department or SIP service provider for the SIP Server configuration, user name and password.


1.5 Configuring Wi-Fi Access Points

Information

The Media5-fone requires the availability of a Wi-Fi network to work properly. Therefore, you need to create one or more proper Wi-Fi access points.

Note: When no configured access points are available, the 3G/4G network of the telephony provider will be used if the Client's settings allow it.

Steps

- 1) From the device's home screen, tap .
- 2) Tap **Wi-Fi**.
- 3) Tap the access point to use and enter the password if prompted to.

1.6 Modifying Your Login Password

Information

You cannot change your login password.

2 Contacts and Contact Lists

There are two types of contacts:

- Network contacts: contacts provided from the RCS Network.
- Native contacts: contacts imported from your device.

There are four types of contact lists:


Contact List Name	Description	Contact List Type
Network or All Network	Contacts provided from the RCS Network.	Predefined
Groups	Contacts selected from the RCS network contacts list.	User defined
Native	Contacts provided from the device (for example by Outlook)	Predefined
All Contacts	Combines both All Network and All Native	Predefined

You cannot directly modify the candidate's profile card. The idea is for the user to own his information and make sure the information displayed to other contacts is approved.

However, for your personal use, you can add information on a candidate, provided the optional feature has been enabled by your administrator. In this case, the system will not modify the candidate's profile but rather create a duplicate of the profile that you will be able to modify. The duplicate will only be visible by you, from all your devices. The added information will not be visible by other contacts.

2.1 Adding a New Contact

Steps

- 1) Tap .
- 2) Tap **Add**.
- 3) Enter a phone number.
- 4) Tap **Done**.
- 5) Tap **Create new contact**.
- 6) In the **Name**, enter the name of the contact.
- 7) Add any required information you have in the contact's fields. Do not hesitate to add another field

- 8) Tap **Done**.

2.2 Adding a Contact via the Dialler

Steps

- 1) From the navigation bar, tap ☰.
- 2) Dial the number of the contact.
- 3) Tap +.
- 4) Tap **Create new contact**.
- 5) Complete the fields as required.
- 6) Tap **Done**.

2.3 Changing the Filter of the History Log

Steps

- 1) Tap ☰.
- 2) Tap **Filter by**.
- 3) In the **Filter by** pop up, select the filter you wish to apply to the history log.

Result

The entries displayed in the history log will correspond to the selected filter.

2.4 Viewing the Communication History Log

Steps

- Tap ☰.

2.5 Deleting the History Log

Steps

- 1) Tap ☰.
- 2) Select **Clear**.
- 3) In the **Clear** pop up, confirm deletion.

Note: Deletion is permanent.

Result



The entries displayed in the history log corresponding to the applied filter, will be permanently deleted. Therefore, if the applied filter is **Declined**, only the **Declined** entries will be deleted.

3 Speed Dialing

3.1 Setting Up a Contact for Speed Dialing


Speed dialing allows you to call a contact with the touch of one single number.

Steps

- 1) Tap .
- 2) Tap ☆ **Favorites**.
- 3) Tap one of the undefined .
- 4) Scroll your list of contact, and tap the contact you wish to associate the icon to.


3.2 Modifying the Number of a Favorite Contact

Steps

- 1) Tap .
- 2) Tap ☆ **Favorites**.
- 3) Tap **Edit**.
- 4) Tap and hold one of your favorite contacts.
- 5) Select the number you wish to use to speed dial this favorite contact.
- 6) Tap **Done**.

3.3 Removing a Contact from Speed Dialing


Steps

- 1) Tap .
- 2) Tap ☆ **Favorites**.
- 3) Tap the contact you wish to remove.

- 4) Tap **Delete**.

3.4 Calling a Favorite Contact (Speed Dialing)

Steps

- 1) Tap .
- 2) Tap ☆ **Favorites**.
- 3) Tap the contact you wish to call.



4 Audio Calls

4.1 Calling with the Dialler

Information


If you know the number to call, you may want to directly use the dialler.

Steps

- 1) From the **Navigation Bar** bar, tap .
- 2) Tap the phone number.
- 3) Tap .

4.2 Calling a Contact

Steps

- 1) Tap .
- 2) In the **Search Contact** field, enter the name of a contact.
- 3) Tap the name of the contact.

4.3 Ending a Call

Before you start

The **End Call** button is only available during an active call.

Steps

- Tap  (**End Call**).

Result

The communication will be ended.

4.4 Putting a Call on Hold

Steps



- 1) Tap .
- 2) Tap  again, to reactivate the communication.

Result

The communication is temporary stopped. You will no longer hear your contact, nor will your contact hear you.

4.5 Putting a Call on Mute

Steps

- 1) Tap .
- 2) Tap  again to reactivate sound.

Result


You will continue to hear your contact, but your contact will no longer hear you.

4.6 Transferring a Call to Another Contact

Before you start

A call must be ongoing. This feature is optional.

Steps

- 1) Tap .
- 2) Select **To Other**.
- 3) From the **Choose Transfer Target** list, select a contact.
- 4) From the **Select a Number** dialog box, select a number.

Result

The call will be put on hold until the call is accepted by the other contact.

4.7 Making a Second Call

Information

A call must be ongoing. This feature is optional.

Steps

- 1) Tap +.
- 2) Dial the number you wish to reach.

4.8 Merging Calls

Before you start

This feature is available provided you can make a second call. You must be in a call for which you were not the initiator.

Steps

- 1) Tap +.
- 2) Dial the number you wish to reach.
- 3) Tap ➤.

Result

A conference call will be created.

4.9 Swapping from One Call to Another

Information

You must have two active calls.

Steps


Tap .

Result

You will be able to have two calls at a time, and swap from one call to another.

4.10 Checking Your Voicemails

Steps

- 1) Tap .
- 2) Tap **Connect to voicemail**.
- 3) Tap the voicemail you wish to listen to.

5 Settings

5.1 Setting the Language of Media5 Client

To change the language of your Media5 Client, change the language of your device. Refer to your device's user guide.

5.2 Changing the Ringtone

Steps

- 1) From the **Navigation Bar**, tap **:**.
- 2) Tap **Settings**.
- 3) From the **Application Settings** section, tap **Ringtone**.

5.3 Enabling Notifications

Steps

- 1) From the **Navigation Bar**, tap **👤**.
- 2) Tap **Settings**.
- 3) From the **Notifications** section, select the notifications you wish to receive.

5.4 Enabling Call Waiting

Steps

- 1) Tap **⋮**.
- 2) Tap **Settings**.
- 3) From the **Services** section, tap **Calls**.

- 4) Check **Call Waiting**.

5.5 Removing a Contact from the Black List

Before you start

The contact must have been deleted from the contact list.

Steps

- 1) Tap ⓘ.
- 2) Tap **Settings**.
- 3) From the **Services** section, tap **Presence**.
- 4) Tap **Blacklist**.
- 5) From the **Blacklist** section, select a number.
- 6) Slide the phone number and tap **Done**.

Result

Your personal information will be visible to the user even if the number no longer appears in your Contact lists.

5.6 Blocking Contact Numbers

Before you start

The contact must have a phone number.

Steps

- 1) From the **Navigation Bar**, tap ☰ .
- 2) Tap **Settings**.
- 3) From the **PRIVACY** section, tap **Blocked Numbers**.
- 4) Tap **Add**.
- 5) In the **Number** field, enter the phone number you wish to block.
- 6) Tap **Done**.

Result



You will no longer be notified of calls and/or chats and/or file transfers from the chosen number, depending on the selections you have made. However they will appear in the **Recents** list.

5.7 Modifying the Speaker Volume






Speaker volume is modified through your device's settings. Refer to the user guide of your device.

5.8 Purchasing Features

Steps

- 1) Tap 
- 2) Tap .
- 3) Tap the feature you wish to purchase.
- 4) Tap **Buy Now**.
- 5) Complete the online instructions.

5.9 Settings for iOS Devices

Task	Description	Setting Location
Auto. Gain Control	To automatically adjust the volume of the microphone.	 / Settings/Media Configuration
Blocked Numbers.	To specify phone numbers for which you do not wish to receive either calls, chat and/or file transfers.	 // Settings/PRIVACY
Call Waiting	To allow a user to receive a call during a call.	 // Settings/Services/Calls
Cellular Data	To enable the use of a cellular data connection. The default value is enabled.	 // Settings/Application Settings
DTMF Sounds	To choose to whether or not play a sound when pressing a digit in the main dialer. The	 / Settings/Application Settings

Task	Description	Setting Location
	mute functionality will override this feature.	
Echo Cancellation	To improve sound quality by suppressing echo during transmission. The default value is disabled.	... /Settings/Media Configuration
joyn	Discover the capabilities of contacts that do not advertise their presence. This can include native contacts.	... /Settings/APPLICATION SETTINGS
Noise Suppression	To suppress background noise to improve the call quality of the application. (Useful if two callers are using the hands-free feature). The default setting is Medium (10dB attenuation).	... //Settings/Media Configuration
Notifications	To choose the notifications you wish to receive. <ul style="list-style-type: none"> • Voicemail • Registration • Missed call 	... /Settings/Notifications
Quality of Service	To make sure VoIP voice packets receive the preferential treatment they require. This could, in some cases, increase voice quality.	... /Settings/Media Configuration
Ringtone	To choose the ringing tone of your device.	... /Settings/Application Settings
Run in Background	To keep the application running even if another application is opened. The default value is enabled.	... /Settings/Application Settings
Starting Screen	To choose the screen that will be displayed when the application is opened.	... /Settings/Application Settings

6 Account Settings

6.1 Account Settings

Parameter	Description	Path
+sip.instance	Defines whether or not to enable the instance ID. This instance ID is used to uniquely identify this specific user on a specific device. This parameter is for advanced users only.	... /Settings/Configure SIP Accounts/name of contact/Advanced/OTHER
3G Codecs	List of active and inactive Codecs available for 3G calls, in order of preference.	... /Settings/Configure SIP Accounts/name of contact/Advanced/MEDIA OPTIONS/Codec
3G Interval (sec)	Sets the interval, in seconds, at which Keep Alive requests are sent to verify the server status via UDP.	... /Settings/Configure SIP Accounts/name of contact/Advanced/Network Options/SIP Keep Alive/UDP Keep Alive
3G Interval (sec)	Sets the interval, in seconds, at which Keep Alive requests are sent to verify the server status via TCP/TLS.	... /Settings/Configure SIP Accounts/name of contact/Advanced/Network Options/SIP Keep Alive/TCP/TLS Keep Alive
3G Keep Alive	To enable or disable the 3G Keep Alive mechanism allowing the server to send messages periodically to ensure that it can still be reached.	... /Settings/Configure SIP Accounts/name of contact/Advanced/Network Options/SIP Keep Alive
Address	SIP server IP address or domain name. This is mandatory. E.g. sip.myprovider.com.	... /Settings/name of contact/Servers
Address	Proxy server IP address or FQDN. Available only if Enable Proxy is checked.	... /Settings/name of contact/Servers/OUTBOUND PROXY

Parameter	Description	Path
Address Mode	<p>Defines how the SDP connection address is modified when holding a call.</p> <ul style="list-style-type: none"> • No change: The SDP connection address is not updated when sending a hold request. • 0.0.0.0: The SDP connection address is set to 0.0.0.0 as defined in RFC 2543. <p>This parameter is for advanced users only.</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Hold and Resume/Hold
Add Transport=tcp	<p>Only available if TCP is selected as the SIP Transport. When set to On, SIP messages are sent with the "transport" parameter in the Contact header of REGISTERS and INVITES set to the TCP transport type.</p> <p>This parameter is for advanced users only.</p>	... / Settings/... /name of contact/ Servers/TRANSPORT AND SECURITY
Allow second call hold	<p>Defines whether or not the Client allows putting a second call on local hold.</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Telephony Services
Auth. Name	<p>Authentication name to use when authentication is required.</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/User Account
Bandwidth Modifier	<ul style="list-style-type: none"> • AS (App Specific Max): Specifies whether or not the "AS" (Application Specific) bandwidth must be sent in an outgoing SDP packet. • TIAS (Transport Independent AS): Specifies whether or not the "TIAS" (Transport Independent Application Specific) bandwidth must be sent in an outgoing SDP 	... / Settings/Configure SIP Accounts /name of contact/ Advanced/MEDIA OPTIONS

Parameter	Description	Path
	<p>packet. Please note that the RFC 3890 specification indicates that the "MaxPRate" attribute must be present if the "TIAS" attribute is present. The specification also recommends that the "AS" bandwidth be present if the "TIAS" attribute is present. Please note that the RFC 3890 specification recommends that the "AS" bandwidth be present if the "TIAS" attribute is present.</p> <ul style="list-style-type: none"> • MaxPRate (Max Packet Rate): Specifies whether or not the 'maxprate' attribute must be sent in an outgoing SDP packet. Please note that the RFC3890 specification indicates that the "MaxPRate" attribute must be present if the "TIAS" attribute is present. <p>This parameter is for advanced users only.</p>	
Complete Caps in Offers	<p>Sets whether or not the complete set of media engine capabilities is to be used to generate an outgoing offer.</p> <ul style="list-style-type: none"> • When enabled, the complete set of media encoding caps is used to generate the media announcement for each media. • When disabled, only the previously negotiated set of media encoding caps are used to generate the media announcement for each media. However, in this mode, the complete set is still sent in offers generated 	<p>.../Settings/Configure SIP Accounts/name of contact/Advanced/Media Capabilities</p>

Parameter	Description	Path
	<p>following the reception of an empty INVITE request.</p> <p>This parameter is for advanced users only.</p>	
Connection Reuse (RFC5923)	<p>Defines whether or not the Client should reuse the TCP/TLS connection (alias parameter in Via header/RFC5923). This setting is only valid when the SIP transport is TCP or TLS.</p> <p>This parameter is for advanced users only.</p>	... / Settings /name of contact/ Servers/TRANSPORT AND SECURITY
Dial+as	Replaces the + with the specified character (must leave empty to remove the +).	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Dial Rules
Direction Mode	<p>Defines how the SDP direction attribute is modified when sending a hold request.</p> <ul style="list-style-type: none"> • No Change: The SDP direction attribute is not updated. • RFC3264: The SDP direction is set to: <ul style="list-style-type: none"> # a=sendonly for a media that has sending capabilities, or # a=inactive otherwise. • Inactive: The SDP direction is set to <ul style="list-style-type: none"> # a=inactive. • Remote Hold Status: The SDP direction is set to <ul style="list-style-type: none"> # a=sendonly when no remote hold is detected for a media that has sending capabilities, or # a=inactive otherwise. 	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Hold and Resume/Hold

Parameter	Description	Path
	This parameter is for advanced users only.	
Direction Mode	<p>Defines how the SDP direction attribute is modified when resuming a call.</p> <ul style="list-style-type: none"> • No change: The SDP direction attribute is not updated. • RFC3264: The SDP direction is set to <ul style="list-style-type: none"> # a=sendrecv for a media that has both sending and receiving capabilities # a=sendonly for media with sending capabilities only # a=recvonly for media with receiving capabilities only # a=inactive otherwise • Remote Hold Status: The SDP direction is set to a=recvonly when no remote hold is detected for a media that has receiving capabilities. Otherwise, the direction is set according to RFC 3264 <p>This parameter is for advanced users only.</p>	... /Settings/Configure SIP Accounts/name of contact/ Advanced/Hold and Resume/Resume
Display Name	Text used as the caller ID. This parameter may be overridden by your SIP Provider / Server.	... /Settings/Configure SIP Accounts/name of contact/ Advanced/User Account
Empty Auth. Header Name	An empty Authorisation header needs to be sent during initial registration procedures in IMS networks. The username in such a header is the user private identity, which is not used anywhere else.	... /Settings/Configure SIP Accounts/name of contact/ Advanced/User Account

Parameter	Description	Path
	This parameter is for advanced users only.	
Enable MKI	<p>Defines whether or not the remote SRTP peer supports the MKI. When enabled, the SDP Crypto attribute is configured with the MKI value of the key and its length. This parameter is only available when the SIP transport is TLS.</p> <p>This parameter is for advanced users only.</p>	... / Settings /name of contact/ Servers/TRANSPORT AND SECURITY
Enable Prack	<p>Defines whether or not the Client supports reliable provisional responses (PRACK) as per RFC 3262. You can define this support when acting as a user agent client and when acting as a user agent server.</p> <p>The Client supports the UPDATE as per RFC 3311; support is however limited to reception.</p> <p>This parameter is for advanced users only.</p>	... / Settings /name of contact/ Servers/TRANSPORT AND SECURITY
Enable Proxy	Defines whether or not the proxy server is enabled.	... / Settings /name of contact/ Servers/OUTBOUND PROXY
Enable SRTP	<p>Defines whether to disable SRTP or enable it with SDES. Please note that setting the SIP Transport parameter to TLS automatically enables SRTP, while setting it to TCP or UDP automatically disables it.</p> <p>Note: If you enable SRTP and you receive a call that is not encrypted with SRTP, a message is displayed asking whether or not you want to accept the call, even though this would result in a non-secured communication.</p>	... / Settings /name of contact/ Servers/TRANSPORT AND SECURITY

Parameter	Description	Path
Force SIP Scheme	<p>Only available if you have selected TLS as SIP Transport. It forces the Client to use SIP instead of the secure SIPS. This may be used for older servers that require unsecure SIP even if the connection uses the secure TLS transport protocol.</p> <p>This parameter is for advanced users only.</p>	... / Settings /name of contact/ Servers/TRANSPORT AND SECURITY
Force Update With Max. Payloads	<p>Sets whether or not an update to the session is required to limit the number of payload types to the maximum number of media payload types specified in the configuration. Following an outgoing offer, if the received answer included more matches than the configured value, then the session is updated with only the negotiated most preferred ones included, up to the maximum specified.</p> <p>This option is ignored if the Complete Caps in Offers setting is enabled or if the Max. Payload Types parameter is set 0, which means no limit.</p> <p>This parameter is for advanced users only.</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Media Capabilities
G.729 VAD	<p>Available only if you have purchased the G.729 Codec. Defines whether or not the G.729 VAD (Voice Activity Detection) feature is enabled.</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/MEDIA OPTIONS
Hold Method	<p>The SIP method to use when performing a Hold or Resume operation. The following methods are available:</p> <ul style="list-style-type: none"> • INVITE • UPDATE 	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Hold and Resume

Parameter	Description	Path
	This parameter is for advanced users only.	
iLBC Mode 20	Defines whether or not iLBC mode 20 is enabled which specifies how much of audio data is sent in each RTP packet. The iLBC codec has two modes (30 [default] and 20). Mode 30 uses 30 ms and 20, 20 ms. If you get bad audio with iLBC, try activating this parameter and see if it makes a difference.	... /Settings/Configure SIP Accounts/name of contact/ Advanced/MEDIA OPTIONS
Local SIP Port End	Ending port used to define the port range used to receive SIP packets. If you leave the fields empty, the Media5-fone uses a default range of 5060 to 5070.	... /Settings/Configure SIP Accounts/name of contact/ Advanced/Network Options
Local SIP Port Start	Starting port used to define the port range used to receive SIP packets. If you leave the fields empty, a default range of 5060 to 5070 is used.	... /Settings// ... /Configure SIP Accounts/name of contact/ Advanced/Network Options
Max. Payload Types	Sets the maximum number of media payload types allowed in a resulting negotiation. For outgoing answers, if the negotiation would produce more matches, then only the most preferred ones will be included, up to the maximum specified. For audio media, CN and NTE encodings are considered as exceptions and are not included in the count. Therefore, even if the maximum is set to one, it could be possible that an answer be generated not only with the preferred audio encoding, but also including CN and/or NTE. It is always assumed that CN and	... /Settings/Configure SIP Accounts/name of contact/ Advanced/Media Capabilities

Parameter	Description	Path
	<p>NTE will be specified last in the media encoding caps. Available values are from 0 (which means no limit) to 65535.</p> <p>This parameter is for advanced users only.</p>	
Method	<p>Allows the Client to send DTMF digits using either:</p> <ul style="list-style-type: none"> • SIP INFO • RFC 2833/4733 RTP Named Telephony Events • RTP Inband 	.../Settings/Configure SIP Accounts/name of contact/Advanced/DTMF
MMTel	<p>Defines whether or not to add the 3GPP MMTEL ICSI in the Contact header.</p> <p>This parameter is for advanced users only.</p>	.../Settings/Configure SIP Accounts/name of contact/Advanced/Feature Tag
NTE Payload	<p>SDP payload for the RFC 2833/4733 DTMF Method, (the number that identifies packets as being RFC 2833/4733 DTMFs). This number must be between 96 and 125. Do not use the numbers 100 to 103 because they are used as follows:</p> <ul style="list-style-type: none"> • 100: Enhanced G.711 ulaw • 101: Enhanced G.711 alaw • 102: iLBC • 103: iSAC <p>The default value is 125.</p>	.../Settings/Configure SIP Accounts/name of contact/Advanced/DTMF
Number	Phone extension used by the Media5-fone to contact your voice mail box.	.../Settings/.../Configure SIP Accounts/name of contact/Advanced/Voicemail
P-Associated Processing	Defines whether or not to enable P-Associated Processing. When enabled, the P-Associated-Uri header(s),	.../Settings/Configure SIP Accounts/name of contact/Advanced/OTHER

Parameter	Description	Path
	<ul style="list-style-type: none"> If it exists in the 2xx REGISTER response, will be processed so that the first header value will change the current user address and each remaining header value will be added to the associated URIs. If it does not exist, the TO header URI will instead be used to change the user address. <p>This parameter is for advanced users only.</p>	
Password	Password used for authentication.	.../Settings/ Configure SIP Accounts/name of contact
Port	Proxy server port number. Available only if Enable Proxy is checked.	.../Settings/Configure SIP Accounts/name of contact/Servers/OUTBOUND PROXY
Port	SIP server port number. The default value is 5060.	.../Settings/Configure SIP Accounts/name of contact/Servers
Prefix to Add	Inserts the specified prefix at the beginning of the string.	.../Settings/Configure SIP Accounts/name of contact/Advanced/Dial Rules
Prefix to Remove	Removes the specified prefix.	.../Settings/Configure SIP Accounts/name of contact/Advanced/Dial Rules
Reg Timer (sec)	<p>Configures the expiration time spent to request in SIP Register requests.</p> <p>It is possible for a server to reject the proposed registration expiration time if it is too short, in which case the shortest possible value supported by the server is used.</p> <p>It is possible for a server to accept the registration while reducing the expiration time,</p>	.../Settings/name of contact/Servers/Other

Parameter	Description	Path
	<p>in which case the registration based on the server provided expiration time is refreshed.</p> <p>The Registration timer parameter has a significant impact on battery lifetime. See Battery Optimization for more details.</p> <p>This parameter is for advanced users only</p>	
Remove Route Header	To send requests directly to the proxy without adding a route header. This fixes interoperability issues with some VoIP providers. Note that this bypasses the standard routing as specified by the SIP protocol. Available only if Enable Proxy is checked.	... /Settings/name of contact/ Servers/OUTBOUND PROXY
RTP Keep Alive (sec)	Sends an empty UDP packet to force firewalls to remain open.	... /Settings/Configure SIP Accounts/name of contact/Advanced/Network Options
RTP Port End	Specifies the port range to use for incoming RTP streams.	... /Settings/... /Configure SIP Accounts/name of contact/Advanced/Network Options
RTP Port Start	<p>The port selection starts at "RTP ports Start" and stops at "RTP ports End" minus "RTP ports Start" divided in 2. This is because the RTCP port is automatically allocated as being RTP Port + 1.</p> <p>Therefore, the NAT/Firewall configuration needs to take this into account and make sure to include "RTP ports End" + 1 as an open port.</p> <p>The range must be sufficient to cover a possibility of 4 simultaneous and instantaneous RTP streams. This allows all call scenarios (2nd call, call waiting,</p>	... /Settings/... /Configure SIP Accounts/name of contact/Advanced/Network Options

Parameter	Description	Path
	<p>transfer, and conference) to work properly.</p> <p>There is no verification on whether or not the range is of sufficient length.</p> <p>However, some call scenarios will not work if the range is not long enough.</p> <p>There is no verification as to if the end port is less than the start port.</p> <p>If the fields are left empty, the default range of 10000 to 10050 is used.</p>	
Service Route Discovery	<p>Enables or disables the use of the SIP Service-Route header:</p> <ul style="list-style-type: none"> • Disabled: Ignores the SIP Service-Route headers present in the SIP REGISTER 200 OK response (if any). • Replace route: Uses the SIP Service-Route header present in the SIP REGISTER 200 OK response (if any) to replace the configured proxy for this user. • Append: Appends the SIP Service-Route header present in the SIP REGISTER 200 OK response (if any) to the configured proxy for this user. 	... / Settings /... /name of contact/ Servers/Other
Session Refresh Mode	<p>Sets the call session refresh mode</p> <ul style="list-style-type: none"> • Treat as new offer/answer • Try to reuse previous SDP • Always reuse previous SDP 	... / Settings/Configure SIP Accounts /name of contact/ Advanced/MEDIA OPTIONS
Session Timer	<p>Enables and configures the Session Timer in seconds. This is</p>	... / Settings/Configure SIP Accounts /name of



Parameter	Description	Path
	used in SIP Calls. Possible values are from 1 to 65535 seconds. This parameter is for advanced users only.	contact/ Advanced/MEDIA OPTIONS
Sip Transport	Defines the transport protocol to use when sending and receiving SIP messages. <ul style="list-style-type: none"> • UDP • TCP • TLS (optional) <p>Note that only TLS is a secured transport protocol. When selecting TLS, the Client automatically switches to the SRTP protocol and uses the SDES key management protocol.</p>	... / Settings/name of contact/Servers/TRANSPORT AND SECURITY
Subscribe MWI	Allows sending a SUBSCRIBE message for the voicemail. <ul style="list-style-type: none"> • ON a SUBSCRIBE message is sent • OFF No SUBSCRIBE message is sent 	... / Settings/Configure SIP Accounts/name of contact/Advanced/Voicemail
TCP/TLS Battery Optimization	Only available if TCP or TLS is selected as the SIP Transport is selected. It allows you to improve your iOS device's battery lifetime. To use this feature, you must also: <ul style="list-style-type: none"> • Raise the Registration timer value to 650 seconds or more. • Disable the Wi-Fi and 3G Keep Alives or lengthen the available intervals 	... / Settings/Configure SIP Accounts/name of contact/Servers/TRANSPORT AND SECURITY
Tel URI	Defines whether or not to enable the conversion of TEL URI to SIP URI when sending a request.	... / Settings/Configure SIP Accounts/name of contact/Advanced/OTHER

Parameter	Description	Path
	This parameter is for advanced users only.	
Title	Name of the account.	... / Settings /... /name of contact
Transfer Method	Defines how the Client initiates transfers. <ul style="list-style-type: none"> • REFER • BYE • Hold and REFER 	... / Settings/Configure SIP Accounts /name of contact/ Advanced/SIP Features
Unattended transfer	Enables or disables the automatic transfer option (unattended transfer).	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Telephony Services
Username	User name for your account. This is mandatory.	... / Settings /name of contact
Use rport	Allows the use of the <i>rport</i> SIP VIA header parameter.	... / Settings /name of contact/ Servers/Other
Wi-Fi Codecs	List of active and inactive Codecs available for Wi-Fi calls, in order of preference.	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Codec/MEDIA OPTIONS
Wi-Fi Interval (sec)	Sets the interval, in seconds, at which Keep Alive requests are sent to verify the server status via UDP.	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Network Options/SIP Keep Alive/UDP Keep Alive
Wi-Fi Interval (sec)	Sets the interval, in seconds, at which Keep Alive requests are sent to verify the server status via TCP/TLS.	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Network Options/SIP Keep Alive/TCP/TLS Keep Alive
Wi-Fi Keep Alive	To enable or disable the Wi-Fi Keep Alive mechanism allowing the server to send messages periodically to ensure that it can still be reached. <ul style="list-style-type: none"> • 3G Keep Alive <p>You can also set the interval, in seconds, at which Keep Alive requests are sent to verify the</p>	... / Settings/Configure SIP Accounts /name of contact/ Advanced/Network Options/SIP Keep Alive

Parameter	Description	Path
	<p>server status. You can set 4 different values:</p> <p>UDP Keep Alive</p> <ul style="list-style-type: none"> • Wi-Fi Interval (sec) • 3G Interval (sec) <p>TCP/TLS Keep alive</p> <ul style="list-style-type: none"> • Wi-Fi Interval (sec) • 3G Interval (sec) <p>The SIP Keep Alive settings have a significant impact on battery lifetime. See Battery Optimization for more details.</p>	

6.2 Modifying Account Settings




Steps

- 1) Tap .
- 2) Tap .
- 3) Tap **Configure SIP Accounts**.
- 4) Tap the account to modify.
- 5) Modify the settings.
- 6) Tap **Done**.

6.3 Creating an Account

this feature is optional and must be purchased to be available.



Steps

- 1) Tap .
- 2) Tap .
- 3) Tap **Configure SIP Accounts**.
- 4) Tap .
- 5) Complete the information

- 6) Tap **Done**.

6.4 Deleting an Account

Steps

- 1) Tap .
- 2) Tap .
- 3) Tap **Configure SIP Accounts**.
- 4) Tap and hold the name of an account.
- 5) Tap **Delete Account**

6.5 Listening to Music while Using Media5-Fone

The Media5-fone fully supports background music playback. If music is playing and you receive or place a call, your music is automatically muted for the duration of the call and resumes once the call terminates.

7 Copyright Notice

Copyright TM © 2016 Media5 Corporation.

This document contains information that is confidential and proprietary to Media5 Corporation.

Media5 Corporation reserves all rights to this document as well as to the Intellectual Property of the document and the technology and know-how that it includes and represents.

This publication cannot be reproduced, neither in whole nor in part, in any form whatsoever, without written prior approval by Media5 Corporation.

Media5 Corporation reserves the right to revise this publication and make changes at any time and without the obligation to notify any person and/or entity of such revisions and/or changes.



4229, rue de la Garlock
Sherbrooke (Quebec)
J1L 2C8 Canada
media5corp.com