

CallMe-T

User Manual

Firmware v2.0



Vortex Communications Ltd, 75 The Grove, Ealing, London W5 5LL, UK

Tel: +44 (0)20-8579 2743 • Fax: +44 (0)20-8840 0018

Email: info@vtx.uk • Web: vtx.uk

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Introduction

CallMe-T is a compact, high-quality stereo IP audio codec with SmartStream™ advanced audio routing and multi-destination streaming. Audio input and output is via ¼" balanced analogue jacks or a USB audio device (headset, mixer, soundcard etc.) attached to the unit's USB 2.0 port. IP connectivity is provided by a 10/100Mbps Ethernet port.

It is available in two models: CallMe-TR and CallMe-TS. CallMe-TS is designed for Studio use, where the unit will primarily be receiving incoming calls and controlled from a computer on the same LAN. CallMe-TR offers all the same functionality as CallMe-TS, with the addition of five front-panel illuminated pushbuttons, and is designed for standalone Remote use.

While CallMe-TS is particularly well-suited as a back end for the Vortex CallMe browser-based contribution service, both models are compatible with a wide-range of existing hardware and software codecs. The units use the industry-standard SIP protocol to establish connections (either via a SIP server or direct to a destination device) and include the Opus audio compression algorithm, as documented in EBU TECH 3326, among others.

To get set up straight away, see Getting Started.

SmartStream™

CallMe-T incorporates SmartStream™ technology. The comprises two components: Advanced Audio Routing and Multiple Simultaneous Calls.

Multiple Simultaneous Calls

When an appropriate licence is installed, CallMe-T can support multiple calls simultaneously (incoming or outgoing, dialled from the front panel or web interface). Up to eight calls are supported at the same time. Each call is independent and can have a different destination, audio coding algorithm and audio routing mode.

Advanced Audio Routing

For each call (incoming or outgoing), it is possible to define how the call's audio is routed to and from CallMe-T's audio outputs and inputs. Three modes are available: Left Only, Right Only and Stereo.

In **Left Only** mode, audio sent to the remote party is sourced from the L input, and the return audio appears on the L output. In **Right Only** mode, the R input and output are used in the same way.

In these modes, when a coding algorithm is selected which offers both mono and stereo modes, CallMe-T will attempt to negotiate a mono connection with the remote party. However, if a stereo connection is established, the send audio will be duplicated from the single mono input across the two encoder channels and the return audio will be mixed down to the single mono output.

In **Stereo** mode, send audio comes from both L and R inputs and return audio is fed to L and R outputs. CallMe-T will attempt to negotiate a stereo connection where the algorithm supports it, but for mono connections the input audio will be mixed down to mono and the return audio will be duplicated across the two outputs.

Audio I/O

As well as the built-in ¼" balanced analogue jacks, CallMe-T supports USB audio class-compliant devices. If such a device is connected when a call is made, it will be used in preference to the analogue jacks. This can be combined with SmartStream™ technology: by making one or more calls, then connecting a USB audio device, then making one or more further calls, it is possible to use the two audio I/O systems simultaneously.

Whenever multiple calls are active and using the same audio output, the audio from the calls will be mixed together.

Auto-Reconnect

When an outgoing call is made on CallMe-T, provided that the call is initially established successfully, the unit will automatically attempt to reconnect if the connection subsequently fails for any reason other than an explicit local or remote hangup (e.g. network failure). This will continue until either it connects successfully or the connection attempt is manually dropped.

Getting Started

Begin by connecting CallMe-T to mains power using the included power supply and to a local Ethernet LAN. CallMe-T is configured and controlled from a web browser.

DHCP Environments

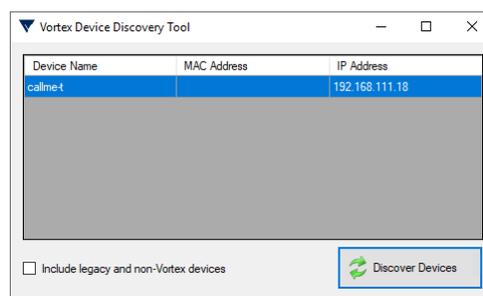
If CallMe-T is connected to a network which incorporates a DHCP server (most networks do, but if you're unsure, consult your network manager), it will be allocated an IP address by the server. In order to browse to it for the first time, it is necessary to know this address. CallMe-T implements the standardized multicast DNS Service Discovery protocol (also known as Bonjour or Avahi) to advertise its web interface on the network, and any mDNS-SD-compatible discovery tool will locate it.

Windows

On Windows, we recommend the Vortex Device Discovery Tool, available from

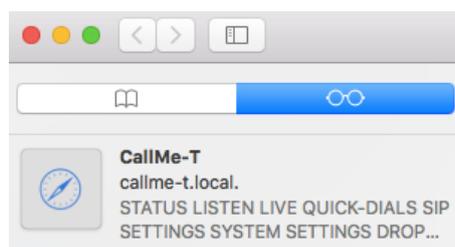
<https://vtx.uk/software/DDT/DDT.exe>

Download and run the application (the [Microsoft .NET Framework 3.5](#) is required). An entry should appear in the list with **callme-t** in the Device Name column; double-click this to open the web interface in a browser.



macOS

On macOS, open Safari, click the "Show sidebar" button if it's not already visible, then click the "spectacles" icon. Click on the callme-t entry which should appear in the list.



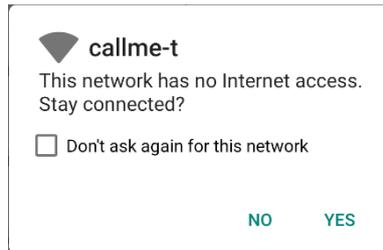
Note that if you have multiple CallMe-T devices with the same name on your network, some will appear in mDNS-SD browsers with numeric suffixes. When setting up, it's recommended that you connect each one to the network in turn and change its name via the web interface (see System Settings) to be able to distinguish them in future.

See Web Interface for further details on logging into and using CallMe-T's browser interface.

In order to facilitate browsing to the unit, you may wish to assign it a static IP address (this applies particularly to CallMe-TS). See Network Settings for further information.

Non-DHCP Environments

If there is no DHCP server on your network to allocate an IP address to CallMe-T, you can use its hotspot mode to access the web interface and configure IP settings manually. Connect a supported USB wifi adapter¹ to CallMe-T's USB port and, from a phone, tablet or computer, look for and connect to an open wifi network named callme-t. The device may notify you that no internet connection is available and ask you to confirm that you wish to stay connected, which you should do.



Open a web browser on the device and navigate to

<https://c.mt/> or <https://10.0.0.1/>

See Web Interface for further details on logging into and using CallMe-T's browser interface and Network Settings for information on setting a static IP.

¹ See Specifications for details of supported adapters.

Front Panel

CALLME-TR ONLY

Connect

Once configured, each of the four CALL buttons (A, B, C and D) on the front panel can be used to dial one of two pre-set destinations (or, when used with CallMe, a “hunting group” in the same way as a CallMe web link).

Pressing one of the buttons briefly will initiate a connection to the “main” quick-dial destination configured for the button. Holding down the DROP button while pressing one of the CALL buttons briefly will initiate a connection to the button’s “alternate” destination (DROP can be thought of as a “shift” key for this purpose). When triggering the alternate destination, take care not to hold the DROP button for longer than 1s without pressing another button as this will drop all calls (see below).

Since CallMe-T is capable of making multiple calls simultaneously², calls can be dialled from the front-panel buttons while others are already active.

Disconnect

To drop a single active call, first press the DROP button briefly; the LED on this button will flash rapidly for 5s while the unit is in “single call drop” mode. During this time, pressing one of the CALL buttons, or holding DROP and pressing one of the CALL buttons, will hang up a call to that button’s main or alternate quick-dial destination respectively (note that selecting a quick-dial to drop in this way involves exactly the same process as selecting one to dial in normal operation).

To drop all active calls (including incoming calls and those which have not been made from the front-panel buttons), press and hold the DROP button for 1s or more.

Status

The LED on each CALL button indicates the connection status of the quick-dials allocated to that button, as indicated in the table below.

The LED on the DROP button flashes slowly to indicate that at least one call is active (regardless of whether it is incoming or outgoing, and whether or not it has been made via the quick-dial buttons).

When CallMe-T is configured to register with a SIP server, the DROP LED will illuminate steadily (whenever a connection is not active) to indicate that the device is successfully registered with the server.

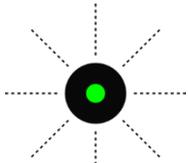
² SmartStream™ – Multiple Simultaneous Calls licence required.

CALL LED



Off

Disconnected



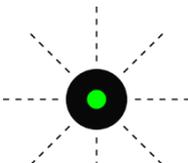
200ms on – 200ms off

Main or alternate quick-dial connecting



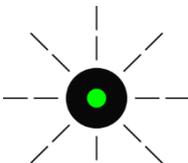
On

Main quick-dial connected only



500ms on – 500ms off

Alternate quick-dial connected only



2000ms on – 500ms off

Both main and alternate quick-dials connected

DROP LED



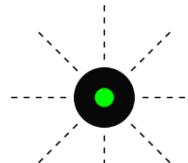
Off

Disconnected
Not registered



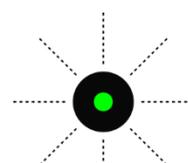
On

Disconnected
Registered



500ms on – 500ms off

At least one call connected



100ms on – 100ms off

Single call drop mode active

Web Interface

CallMe-T can be operated via its built-in web interface, by simply entering the IP address of the unit into the address bar of a browser and pressing enter. It also supports encrypted connections to the web interface by prefixing the unit's IP address with https://

Note that because the unit uses a "self-signed" certificate for https, the browser will present a security warning because it cannot verify the identity of the device. However, the connection will still be encrypted, and in this case it is safe to ignore the warning, as follows:

- Chrome** Click Advanced, Proceed to xxx.xxx.xxx.xxx (unsafe).
- Edge** Click Details, Go on to the webpage (Not recommended).
- Firefox** Click Advanced..., Accept the Risk and Continue.

You will be prompted to log into the web interface. The username is **admin**, and the default password is the 12-character MAC address of the device (shown on a label on the bottom of the unit), in lowercase, without any colons or dashes.

Status

This tab shows a list of any active calls. For each call, the following information is shown.



Connection For outgoing calls, the "friendly name" assigned to the connection (if any).
For incoming calls, the name provided by the calling party.

The SIP URI of the remote party.

Call State The current state of the call: CONNECTING or CONNECTED.

Audio The audio routing and levels for the call.

A L speaker icon and level meters indicates a Left-Only call.

A R speaker icon and level meters indicates a Right-Only call.

Both L and R speaker icons and level meters indicates a Stereo call.

The TX meters measure the audio from each input which is being sent to the remote party.

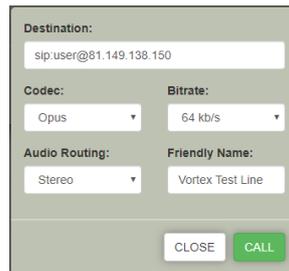
The RX meters indicate the audio from the remote party sent to each audio output.

DROP Click DROP to disconnect the call.

CALL...

To make an outgoing call, click the CALL... button and enter the destination address. Optionally, choose a codec, bitrate (not available with all codecs) and audio routing mode from the drop-down fields. To help identify the call in the calls list, you can provide a name in the Friendly Name field. Click CALL to dial.

The destination address may be either the name of another user on the same SIP server (when registered), or a full SIP URI in the form **sip:user@host**. Here, **user** is the SIP username of the destination (for some direct-dialled devices this may be anything) and **host** is the destination hostname or IP address.

A screenshot of a dialog box titled "Destination:". It contains a text input field with the value "sip:user@81.149.138.150". Below this are two columns of settings: "Codec:" with a dropdown menu showing "Opus", and "Bitrate:" with a dropdown menu showing "64 kb/s". The next row has "Audio Routing:" with a dropdown menu showing "Stereo", and "Friendly Name:" with a text input field containing "Vortex Test Line". At the bottom right, there are two buttons: "CLOSE" and "CALL".

For example, to make a direct SIP call to our test line, enter **sip:user@81.149.138.150**. If registered with CallMe, a call can be made to any public destination by entering the part of the CallMe URL after <https://callme.fm/>). Therefore our test "studio", available at <https://callme.fm/ealing1048>, can be dialled by entering **ealing1048**.

DROP ALL

To disconnect all active calls, click the DROP ALL button.

Quick-Dials

CALLME-TR ONLY

The settings for the four front-panel quick-dial buttons can be configured on this tab. There is a separate group of options for the main quick-dial on each button (e.g. “CALL A”), and for the alternate quick-dial (e.g. “CALL A + DROP”). The options for each are identical to those in the Call dialog on the Status tab; see CALL... for further information.

After making any changes to the quick-dial settings, click the SAVE button at the bottom of the list.

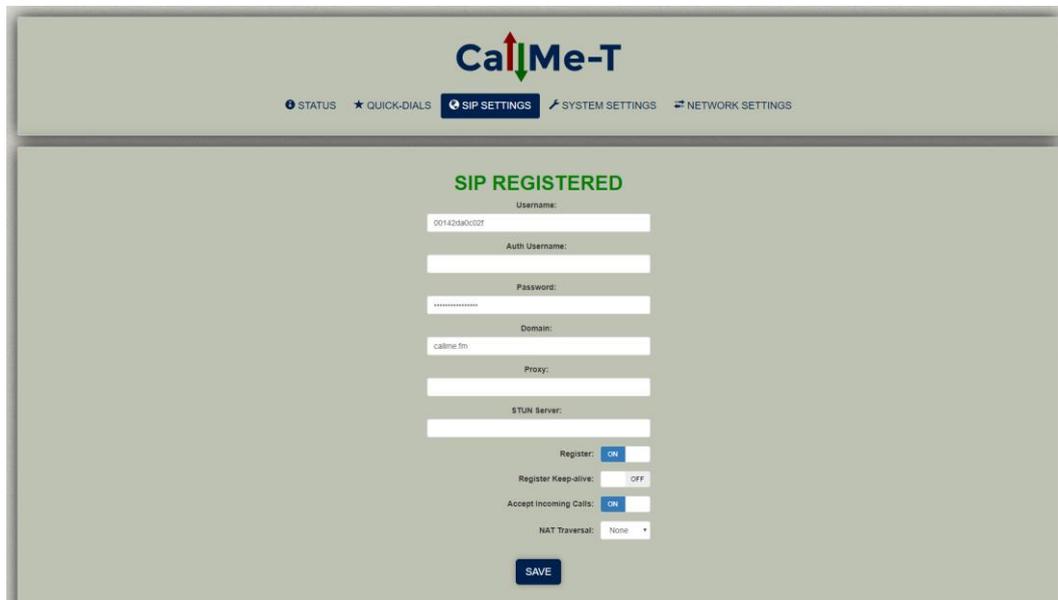
It is possible to connect directly to one of the configured quick-dials by clicking the CALL button on the relevant settings panel. If you are also modifying the settings, we recommend saving the changes first as the CALL button will immediately return to the Status tab.

The screenshot shows the 'Quick-Dials' configuration page for CallMe-T. The page has a header with the CallMe-T logo and navigation tabs: STATUS, QUICK-DIALS (selected), SIP SETTINGS, SYSTEM SETTINGS, and NETWORK SETTINGS. The main content area displays three quick-dial configuration panels, each with a 'CALL' button. The first panel, 'CALL A', has a destination of 'sip:user@81 149 138 150 5060', Opus codec, 64 kb/s bitrate, Stereo audio routing, and 'Vortex Test' as the friendly name. The second panel, 'CALL B', has a destination of 'calling1048', Opus codec, 64 kb/s bitrate, Stereo audio routing, and 'CallMe Demo' as the friendly name. The third panel, 'CALL C', has a destination field that is currently empty.

SIP Settings

This tab determines the details CallMe-T uses to register with a SIP server. In order to register, a minimum of Username, Password and Domain are required. The correct values for all these settings will be specified by your SIP provider.

Click SAVE after changing any settings on this tab.



Username The username used to register with the SIP server. This is the **user** part of the device's SIP URI **sip:user@domain**.

For registrations to CallMe, the username is the device's MAC address, in lowercase, without any colons or dashes.

Auth Username The username to use for authentication with the SIP server. If not provided, defaults to the same value as Username.

Password The password to use for authentication with the SIP server.

Domain The domain at which to register. This is the **domain** part of the device's SIP URI **sip:user@domain**.

For registrations to CallMe, the domain is **callme.fm**.

Proxy The SIP proxy server to use. If not provided, defaults to the same value as Domain.

STUN Server The STUN (and optionally TURN) server to use when NAT Traversal is enabled (see below). Your SIP provider will normally supply a STUN server address, but the Google public server **stun.l.google.com:19302** can also be used (STUN only, not TURN).

Register This should be set to ON in order to register with a SIP server using the details configured above.

Register Keep-Alive Setting this to ON helps to keep ports open through any firewalls or NAT routers between CallMe-T and the SIP server.

This is not required for registrations to CallMe but may improve compatibility with some other SIP servers. If incoming calls fail intermittently, try enabling this option.

Accept Incoming Calls By default, CallMe-T accepts any incoming calls. If you prefer to use CallMe-T only to make outgoing calls to another device (this applies particularly to CallMe-TR), set this to OFF.

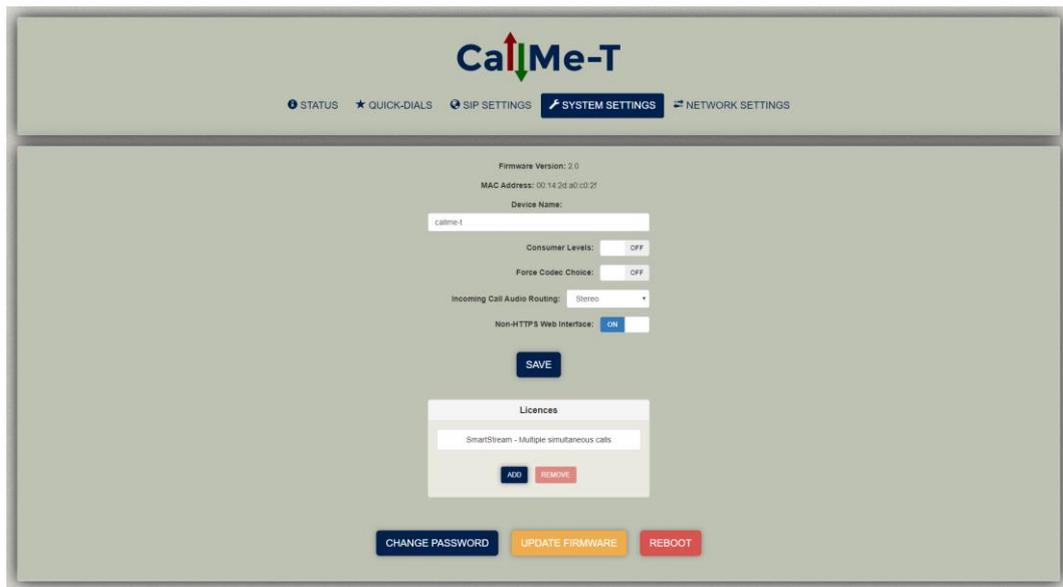
NAT Traversal CallMe-T supports two optional methods of NAT router / firewall traversal: the simpler STUN protocol to determine its public IP address, and the full ICE protocol (including TURN, when used with a suitable server).

Neither is necessary when registering with CallMe (the setting may be left at None), but they may help if you experience connectivity issues with third-party SIP servers or when making direct calls.

If this is set to either STUN or ICE, a STUN (or STUN + TURN) server address must be specified in the STUN Server field.

In general, if your SIP provider has specified a STUN server address, you should set this option to STUN (and enter the address in the STUN Server field); if they have provided a TURN server, you should choose ICE here (and enter its address in STUN Server).

System Settings



Device Name To distinguish multiple CallMe-T devices on the same network when using a mDNS-SD browser, the name with which a unit advertises itself can be changed here. This also determines the name of the hotspot created when a USB wifi adapter is connected.

Consumer Levels By default, CallMe-T operates with professional audio input and output levels (0dBu nominal). To use the device in a consumer environment (for example where the inputs or outputs are interfaced to a computer soundcard or when connected to domestic loudspeakers), set the Consumer Levels switch to ON. This boosts the audio input gain and correspondingly reduces the output level.

This only affects the levels of the on-board analogue audio inputs and outputs and has no effect on USB audio devices.

Force Codec Choice The Codec setting in the Call dialog and on the quick-dial configuration panels specifies the audio coding algorithm which CallMe-T will advertise to the remote party as its preferred choice when a call is negotiated. However, by default it will still offer all other supported algorithms as lower-priority options and the destination may select one of these instead.

Setting this to ON removes these additional codecs, which will ensure that the call will either use the selected codec, or fail.

Incoming Call Audio Routing This setting determines how the audio is routed for incoming calls. Stereo, Left Only and Right Only specify the routing for all incoming calls, while Alternating results in the first call being Left Only, the second call being Right Only, the third being Left Only and so on. For more information about the different modes, see Advanced Audio Routing.

Non-HTTPS Web Interface

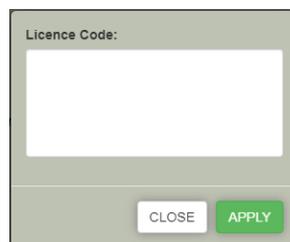
If you wish to force all control connections to CallMe-T's web interface to be encrypted, disable the unencrypted (HTTP) interface by setting this to OFF.

After changing any of these settings, click the SAVE button.

Licences

The Licences panel shows any optional features which have been enabled on CallMe-T. To add a feature for which you have a licence, click ADD, paste the code into the appropriate field and click APPLY. There is no need to reboot for the licence to take effect.

To obtain a licence code from Vortex you will be asked to provide the unit's MAC address; this is shown at the top of the System Settings tab.

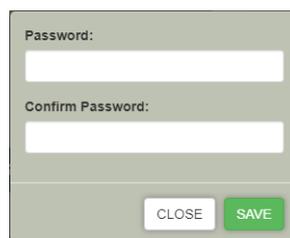


A dialog box titled "Licence Code:" with a large empty text input field. At the bottom right, there are two buttons: "CLOSE" and "APPLY".

To remove a licence (for example, if you need to install a different version of an existing licence), select the licence in the list and click REMOVE.

Change Password

The CHANGE PASSWORD button allows the password for the web interface to be changed; click to display the password dialog, type the new password twice and click SAVE. Note that the username is always **admin**.



A dialog box for changing the password. It has two text input fields: "Password:" and "Confirm Password:". At the bottom right, there are two buttons: "CLOSE" and "SAVE".

Firmware Update

The current version of firmware running on CallMe-T is shown at the top of the System Settings panel; the firmware can be updated by clicking the UPDATE FIRMWARE button and selecting a new firmware file (.fwu). The update process takes about 30 seconds, during which time you should not close your browser.

A message will be displayed below the UPDATE FIRMWARE button once the firmware update has completed successfully, but the new firmware will not take effect until the device is rebooted. This can be achieved using the REBOOT button or simply by power-cycling the unit.

CallMe-T's settings and licences will be retained across a firmware upgrade.

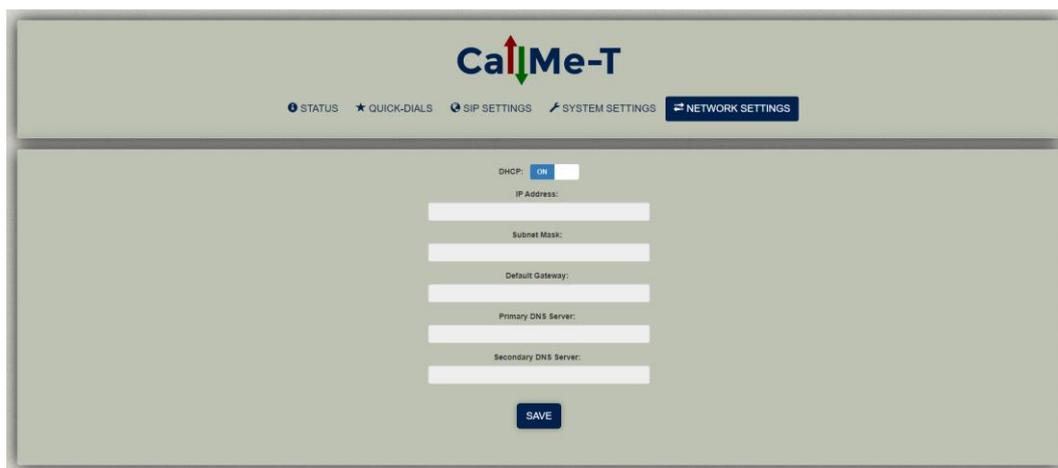
We recommend that you keep your unit up to date. The latest firmware version can be downloaded from <https://vtx.uk/software/CallMe-T>. If you would like to be notified by email whenever a new firmware version is released, please contact info@vtx.uk.

Network Settings

To configure CallMe-T to use a static IP address, set the DHCP switch to OFF and fill in the appropriate information in the fields below it. At a minimum, IP Address and Subnet Mask are required, but Default Gateway is needed to communicate with devices outside the local network and at least one DNS server is required when calling devices or registering with a SIP server using a host name (rather than an IP address). As an alternative to using local DNS servers, the Google public DNS servers **8.8.8.8** and **8.8.4.4** can be used.

You should consult your network administrator to determine the correct settings to use to avoid conflicting with other network devices.

To have CallMe-T obtain its IP settings automatically from a DHCP server on the network (the default behaviour), set the DHCP switch to ON.



The screenshot shows the 'CallMe-T' web interface with the 'NETWORK SETTINGS' tab selected. The interface includes a navigation menu with 'STATUS', 'QUICK-DIALS', 'SIP SETTINGS', 'SYSTEM SETTINGS', and 'NETWORK SETTINGS'. The main content area features a 'DHCP' toggle switch set to 'ON'. Below this are five input fields for 'IP Address', 'Subnet Mask', 'Default Gateway', 'Primary DNS Server', and 'Secondary DNS Server'. A 'SAVE' button is located at the bottom of the form.

Click the SAVE button to commit changes to the network settings. Note that when supplying static IP information, the SAVE button will be disabled if the settings entered are not valid or logically consistent; in this case, please check the values carefully and amend where necessary.

When switching from DHCP to a static IP address, the web interface will attempt to redirect to the new address automatically when the settings are saved. However, on changing to DHCP, it will be necessary to use a mDNS-SD browser to find the new IP address of the device (see Getting Started).

USB Keyboard Support

A keyboard or number pad can be connected to CallMe-T's USB port to make and drop calls.

To dial, type the destination address using the keyboard in exactly the same format as you would in the Destination field of the Call dialog. If CallMe-T is registered with a SIP server, you can type just a username (which will need to be purely numeric if using a number pad); otherwise you should type a full SIP URI in the form **sip:user@host**. If you make a mistake, you can erase previous characters typed using the keyboard's backspace key.

Once you have entered the destination, place the call as follows, depending on the audio routing mode required:

Stereo Press Enter ↵.

Left Only Hold down Enter ↵, press / (on the number pad if using a full keyboard), then release Enter ↵.

Right Only Hold down Enter ↵, press * (on the number pad if using a full keyboard), then release Enter ↵.

To drop all calls, press and hold ESC, or . on the number pad, for 1s or more (ESC and number pad . replicate the DROP button on a CallMe-TR).

Factory Reset

To reset CallMe-T to its factory configuration, press and hold the front-panel DROP button (CallMe-TR only), or the ESC or number pad . key on a connected USB keyboard, for 30s. On a CallMe-TR, the CALL B, CALL D and DROP LEDs will illuminate and then extinguish as the unit reboots at the end of the reset process.

Note that the factory reset process will return the unit's password to its default value (see Web Interface for details) and its network settings to DHCP. Any licences for optional features will be retained across the factory reset.

Specifications

Supported Codecs	Opus Mono and Stereo, 8–510kbps G.722 G.711 μ -law and A-law
Analogue Audio Input	L & R balanced ¼" jacks Level: Line, 0dBu nom., 24dBu max. Impedance: 24k Ω electronic transformer balanced
Analogue Audio Output	L & R balanced ¼" jacks Level: Line, 0dBu nom., 24dBu max. Impedance: 100 Ω electronic transformer balanced
USB	USB host connectivity for use with: <ul style="list-style-type: none">• USB audio class-compliant interfaces• USB keyboards / number pads• Selected USB wifi adapters for local hotspot control (devices based on the Ralink rt33xx or rt35xx chipsets, or the Realtek RTL8192CU or RTL8188CU chipsets, are supported; please consult Vortex for further information)
Analogue Frequency Response	20Hz–22kHz \pm 3dB
Network	100BASE-TX Ethernet on RJ45
IP Protocols and Ports	SIP UDP 5060, TCP 5060 RT(C)P UDP 15004–15515 mDNS UDP 5353 LLMNR UDP 5355, TCP 5355 DHCP UDP 68 DHCPv6 UDP 546 DNS UDP <i>any</i> (53 remote) STUN UDP <i>any</i> (3478 remote) HTTP TCP 80 HTTPS TCP 443 SSH TCP 22
Power	8–26V DC 2.1/2.5mm coaxial connector 24V DC external mains power supply included
Dimensions	108mm x 103mm x 44mm desktop enclosure (extruded) Rackmount: Optional rack adapter (19" x 1-U) holds up to 4x CallMe-Ts
Weight	0.3kg