

MOBImanual

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Mobi basics

Mobi was designed as a remote tool for newsrooms. It enables reporters in the field to create a transmission link to the studio with broadcast quality.

Main features are:

- Opus Codec (RFC 6716), mono
- Adjustable encoding bandwidth
- Level display
- Record and playback functionality

In conjunction with a hardware codec like Mayah C11 or CENTAURI III, and a suitable wireless connection (Wifi, LTE), reporters can go on air simply by using their device.

Mobi uses SIP for communication with the hardware codecs. While SIP is widely deployed, it needs to be carefully set up, especially with regards to NAT traversal. This document describes two possible deployment scenarios.

Mobi Scenarios

Mobi was intended for two main connection scenarios, *direct connection* and with a *SIP-account*.

Direct Connection

In this case, the hardware codec is set to a public IP address, and the Mobi can dial the IP address or domain name of the hardware codec. Here, Mobi does not have to be configured to a SIP account.

This setup is expected to work fine, even when Mobi is behind a Network Address Translation (NAT) router in a Wifi network, because Mobi and the Mayah hardware codecs¹ support symmetric RTP, which will enable bidirectional audio paths in most cases.

If the hardware codec cannot be set to a public IP address - e.g. because no public IP address is available - there are two other options:

Using SIP proxy on the studio gateway

Using a STUN service STUN, RFC 5389, <https://tools.ietf.org/html/rfc5389>

The first option is strongly preferred. STUN works well in many cases, but it may not work with some firewalls. Also, public STUN services may not be highly reliable. But if the firewall works well with STUN, and the STUN servers are reliably reachable, STUN may be an option.

If a SIP proxy is used, it is recommended to use a kernel-level RTP proxy for minimal delay and jitter. See mediaproxy-ng, kernel level RTP proxy, <https://github.com/sipwise/mediaproxy-ng> for an implementation of a kernel-level RTP proxy.

With a SIP-account

In this configuration, the radio station operates a SIP Registrar and Session Border Controller(s), abbreviated SBC(s). Mobi and the hardware codecs are configured to register with the SIP Registrar. In this scenario, the Mobi user dials the extension of the hardware codec.

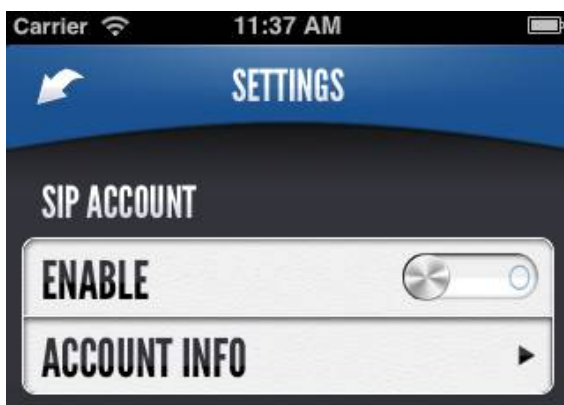
The SBCs are configured to support the devices with NAT traversal. As in the usecase above, it is recommended to run a kernel-level RTP proxy for minimal delay [2].

This option gives best control over the media stream routing, and also enables usecases like automatic recordings in the SBC.

Mobi configuration

SIP Account

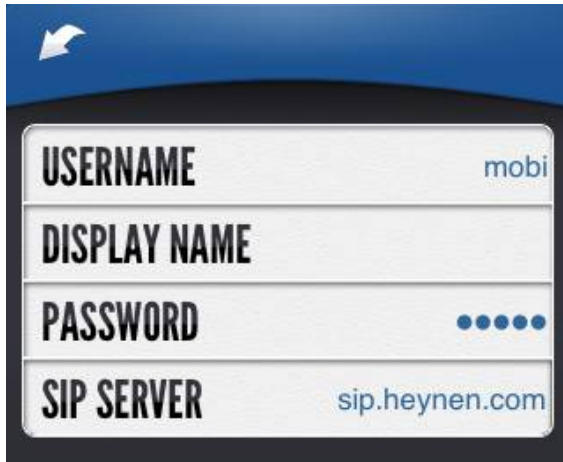
Depending on the scenario, a SIP account may not be necessary. If the scenario *direct connection* is used, the SIP account can set to enable *off*.



If an account is used, it can be configured in the *Account Info* subscreen.

¹

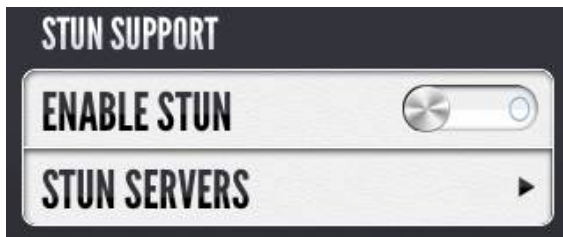
¹ Since version 4.9 of the CIII firmware.



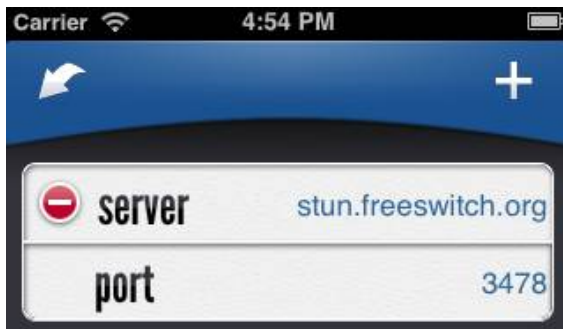
Username, Password and SIP Server are used for authentication with the SIP Registrar. The *display name* is shown to the opposite party. It can be a name or a number.

Stun Support

STUN The Opus audio codec, <http://tools.ietf.org/html/rfc6716> is a NAT traversal support service. Because Mobi and the hardware codecs use symmetric RTP, it should be ok to leave this to the *off* position.



A list of public STUN servers can be found here: <http://www.voip-info.org/wiki/view/STUN>, but please be advised that public STUN servers may not be reliable.



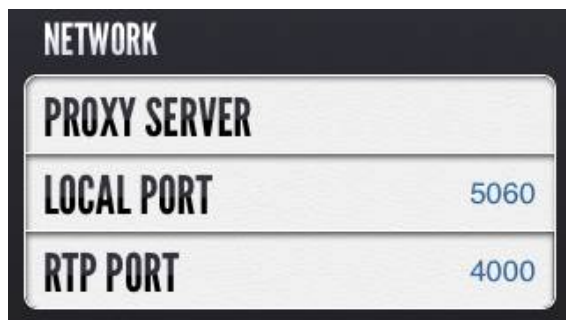
Multiple STUN servers can be configured to improve reliability.

The default port for STUN is 3478. Unless your STUN server uses a nonstandard port, keep the default setting.

Network

With the exception of the proxy server, these settings should not be changed, unless your system administrator requires changes.

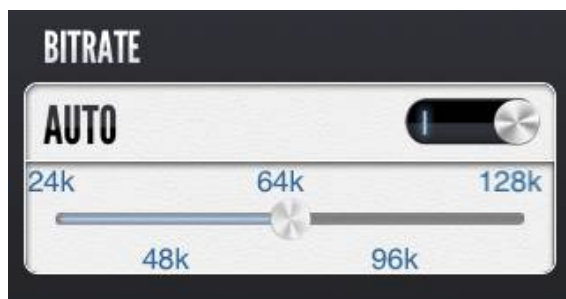
The proxy server can be useful when you are in a closed (Wifi) network which has a SIP proxy.



Bitrate

Changing the bitrate alters the outbound bitrate, i.e. it influences the perceived sound quality for the other party. It is recommended to leave it at the default of *Auto on*.

In general, a lower value will lead to lower sound quality, and higher values run the risk of insufficient bandwidth and dropouts.



Jitter buffer

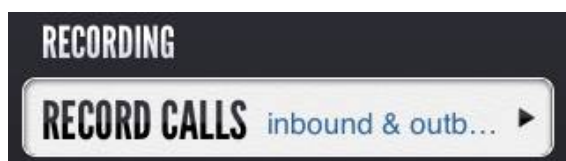
The jitter buffer is needed to smooth out the variable transmission times of audio packets. Larger values make dropouts less likely, but increase the reception delay.

If *adaptive* is set to on, the jitter buffer grows and shrinks with varying network conditions.



Recording

Mobi can record calls. Use *Recording* to specify whether the inbound direction should be recorded, the outbound, or both.



Version & Debug logs

This section shows the version number (*1.0*), build number (*77*) und internal revision number (*bfeb099*).

Send debug logs sends the log files for the last 5 calls per email to mobi@mayah.com. These log files contain the SIP signaling, including names and numbers of called and calling parties, but no recordings of transmitted speech.



Testing

Without any account configuration, Mobi can be tested with the following devices:

62.214.4.53 (echo device)

62.214.4.54 (radio stream)

Known bugs

The adaptive jitter buffer produces audible artifacts when compressing. It is recommended to leave the Jitter buffer setting *adaptive* to off and determine the needed jitter buffer size with test calls.

Please note that the jitter buffer size can not be modified while a call is in progress.

References

- [1] The Opus audio codec, <http://tools.ietf.org/html/rfc6716>
- [2] SIP, Session Initiation Protocol, <http://tools.ietf.org/html/rfc3261>
- [3] STUN, RFC 5389, <https://tools.ietf.org/html/rfc5389>
- [4] mediaproxy-ng, kernel level RTP proxy, <https://github.com/sipwise/mediaproxy-ng>

Addendum

Usage hints for sipgate.de

<http://www.sipgate.de/> uses two usernames passwords. The first pair is used for web login, the second, important one for Mobi is hidden under Settings - *Einstellungen*.

The Mobi username is called *SIP-ID* und the Mobi password is called *SIP-Passwort*.

Einstellungen

Übersicht **Persönliche Daten** **Vertrag** **Telefonie** **Voicemail** **Weiterleitung** **Fax** **Portierung**

SIP-Accountdaten

Diese Accountdaten müssen in der Konfiguration Ihres/Ihrer benutzten Endgeräte eingetragen werden. sipgate stellt zu den einzelnen Endgeräten detaillierte Konfigurations-Anleitungen bereit.

[Weitere Informationen finden Sie hier.](#)

SIP-ID: 1234567
SIP-Passwort: X8FGXZ
Status:  offline
Bezeichnung: kein Name gesetzt [bearbeiten](#)

SIP-Serverdaten

Diese Accountdaten müssen in der Konfiguration Ihres/Ihrer benutzten Endgeräte eingetragen werden. sipgate stellt zu den einzelnen Endgeräten detaillierte Konfigurations-Anleitungen bereit.

[Weitere Informationen finden Sie hier.](#)

Registry: sipgate.de (Port: 5060)
Proxy: sipgate.de (Port: 5060)
NTP: ntp.sipgate.net