# MAYAH Communications
# Application Note 36

# Using the Mayah SIP Server

## 1. General description

MAYAH Communications provides a professional SIP server (available as **sip.mayah.de** starting from 01.04.2015 and as **sip.mayah.com** starting from 01.06.2015).

Please contact us to get your account on our SIP server. This server supports FEC, all kinds of NAT and Interconnect!

It is beyond the scope of this document to give a complete overview over the functionality that SIP provides. Figure 1 gives an overview over the signal flow during a typical SIP Session.

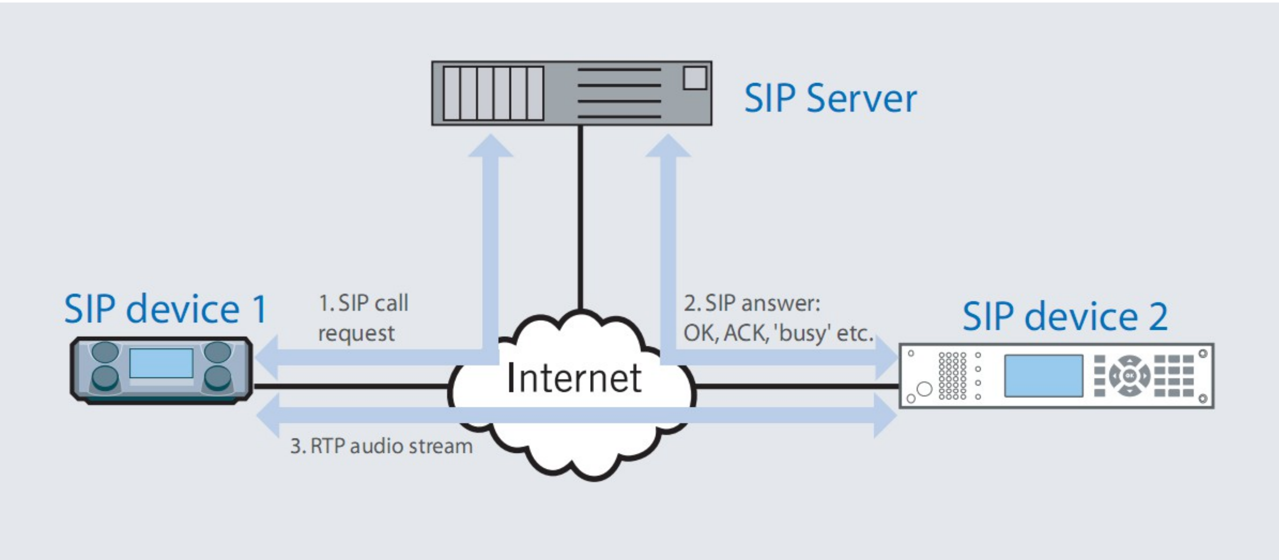



**Figure 1: SIP Signal Flow**

## 2. Prerequisites

In order to use the SIP server you need the following:

- An Internet connection (this could be a 3G or 4G connection as well)
- A vaild DNS Server setting (this is most likely correct if you use DHCP)

## 3. How to enable the SIP account

MAYAH codecs are equipped with a minimum of five registration slots (Account IDs).

Your codec can be registered on every account at the same time.

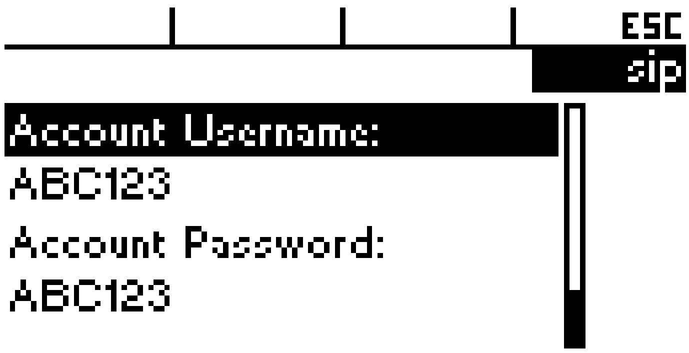
Please deactivate STUN for every account ID if Mayah SIP server is configured.

## 3.1 via Front panel

for the current product family the SIP settings can be found under CODEC -> SETUP -> INTERFACE -> Ethernet->SIP

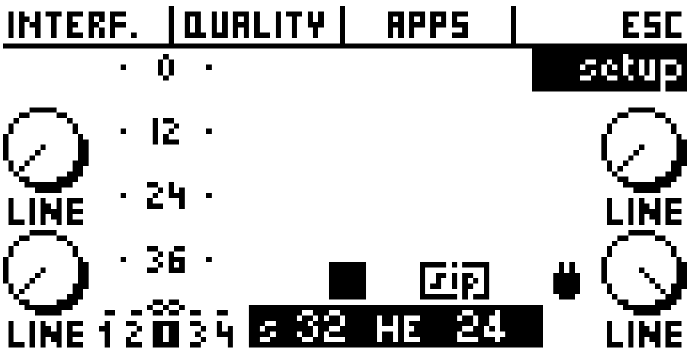
Please select an **Account ID** (registration slot) and make sure that **Account active** is set to **off.** Otherwise the changes you make to the account will not be committed.


Now enter the login details.


Please use the login details as described in Section 1.

Now you can turn on the SIP account in order to register with the server. If successfully registered you will see the SIP logo in your status bar.


### 3.2 via Remote Software

The same settings can be done in the corresponding menus of your remote software. E.g. SETUP/Ethernet/SIP or Settings/SIP)

## 4. Placing calls

To call another codec through the MAYAH SIP Server you need to select SIP as the protocol for the call. If you plan to use a phonebook entry or are using remote commands you need to prefix the destination address with '**sip:**'; for example: '**sip:ABC123'** (without quotation marks)**.**
The destination address must be that of a codec that is already registered to the MAYAH SIP Server (direct peer-to-peer SIP calls are also possible, but not subject of this Application Note). To dial, you can either use ABC123@sip.mayah.com or only ABC123 . Where ABC123 is the Phonenumber from the login details of the peer device. Please note, that the Phonenumber may be case sensitive.
For testing purposes a Codec is available under 'mayah1'. Feel free to use this for tests, but keep in mind, that this is just one codec and other users might be trying to use it.