APPLICATION NOTE

Connecting AEQ Phoenix Mobile units via Internet.

Complex scenario configuration.

AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0-F

Connecting AEQ Phoenix Mobile units via Internet, complex scenario configuration

Through LOCAL network(s), DHCP not used, manual NAT. DIRECT SIP mode. Adapted to Phoenix Mobile User Interface

The most convenient and reliable way to connect an IP codec to a remote one through Internet is by means of a dedicated, exclusive Internet access (DSL, cable modem). In this application note, a way to achieve the same result will be proposed when this is not possible as Internet access is shared with other equipments connected within a managed, LOCAL LAN and DHCP is not available or convenient. This application note describes the recommended procedure when AEQ SIP Proxy is not reachable or we don't want to use it, and SIP signaling is used from end to end, with no servers involved.

As we need to get access to Internet, a shared resource in this case, it is mandatory to contact the network administrator in order to have the required resources enabled and gather all the necessary information.

This document shows the specific steps to make this connection possible in the most general way possible, although some verification procedures have been omitted in order to ease the reading, and some particular situations may be easier to solve that the presented one.

For a detailed explanation of every step, and for details on how to configure other options that may be more convenient for other cases, the reading of the rest of application notes, as well as chapter 4 of user's manual, is recommended.

1. NETWORK ADMINISTRATOR INVOLVEMENT

We will need that the Network Administrator:

1. Provides us with a valid internal IP within the local network, for example 172.26.5.59, for sending SIP signaling as well as audio using RTP protocol. Both IP will be the same in both cases, and we will refer to it as **LOCAL IP**.

2. Indicates us the **Network Mask**, for example 255.255.0.0

3. Tell us the **DNS server IP** address, for example 172.26.1.1 or a valid external DNS server for the country we are working at. If it is not provided, we will still be able to operate provided that we know the IP address of the SIP Proxy (although it is always recommended to specify it by its DNS name)

4. Provides the **Gateway IP address**, as an example 172.26.1.1

5. Indicates us the network **Public IP address**. In most cases, there will be a single public IP for SIP signaling and RTP traffic. We will refer to it as **PUBLIC IP**, for example, 212.170.180.170

6. **Opens the required ports** for audio and signaling. These ports are, by default:

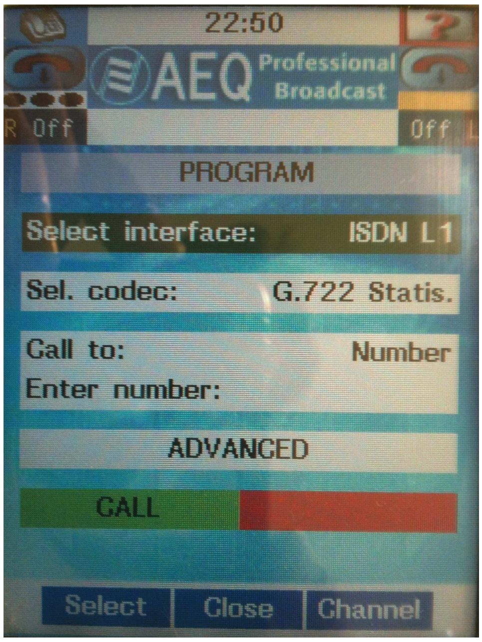
    - SIP signaling PORT (SIP LOCAL PORT): 5060
    - Ports for audio transmission and control (RTP and RTCP protocols): 5004 and 5005, respectively.
    - STUN ports: 3478 & 3479.

2. CONNECTING AND CONFIGURING THE UNIT

The physical connection will be done through a LAN port.

- Connect the network cable to the LAN port at the back of the AEQ Phoenix Mobile. Connect the other end to the Ethernet port to the router LAN port.
- Connect at least a microphone and a pair of headphones to the front audio XLR connectors of the Phoenix Mobile
- Connect the power supply cable to the AEQ Phoenix Mobile. If it is equipped with a battery pack and mains power is not available, you can omit this step.
- Press the ON button at the left side of AEQ Phoenix Mobile. Verify that the main display is illuminated.
- Verify, at the back of the unit, that the green LED at the right of the Ethernet port starts blinking indicating that the IP connection is active.
- Configuring the Ethernet port and assignation of Program channel:

  Make sure that we are selecting "Program" in the main screen (the one representing faders and vumeters). At the lower center area you should read "Program". Also, by pressing MENU we enter the Wizard menu that should have a PROGRAM legend at the top center area. If COORDINATION appears instead in any of both places, just press Channel key and channel will be shifted to PROGRAM.



MAIN SCREEN WIZARD SCREEN

From main screen press Menu to access the Wizard menu. Make sure that Select Interface shows “IP (Ether)”. Otherwise, please select this.

Highlight the <u>Configure Interface</u>, line using the down arrow key, press Select and configure the following:

- DHCP: OFF
- Local IP: type **IP LOCAL** address from 1.1, for example: 172.26.5.59
- Gateway: type the **Gateway** IP address for Internet access, from 1.4. For example: 172.26.1.1
- Mask: type a valid **Net Mask** from 1.2, for example: 255.255.0.0
- DNS Server: type a **DNS IP** address, from 1.3, for example: 172.26.1.1
- **SIP Proxy: OFF**

Make sure that the STUN server is activated and with a valid address set, as explained afterwards.

3. OTHER POINTS TO CHECK

We are now going to quickly overview the rest of configuration menus where it is recommended to check that every parameter is correctly configured.

Coding Algorithm

At the Wizard menu Sel.Codec should be G722, for a start, although you can change this at a later stage for better quality. In order to change the coding algorithm by another pre-selected one, just select the Sel.codec line, press Select and select one within the new window shown. In order to choose the particular coding modes in each list shown in the Sel.codec menu, go to Menu → ADVANCED → 6. Link profiles → Select → Options.

The audio routing will be most likely correct by default to send Mic 1 audio to the transmission and send the received stream to HP1 headphone output. You can, however, make sure of this from Menu → ADVANCED → 3. Audio Settings → Select.

Configure the rest of parameters after reading the manual according to your particular needs.

DIRECT SIP configuration (PROXY SIP OFF)

You can make sure that this is correct from the main screen: Menu → ADVANCED → 5. Communications → 3. SIP config → 1. Parameters

Check the User Name.

STUN Protocol

2 Activate ON (this is important and recommended, as it will ease connecting to other equipment that is behind a router in a local network).

3 Server: At the moment of writing this document, the following STUN server is recommended: **stun.sipgate.net** (IP=217.10.68.152)

If the DNS Server field in the communications menu hasn't been configured, you must substitute the STUN name by its IP.

4 Port: 3478

Press "Back" key in order to go to 3. SIP config → 2. Proxy and check:

1 Activate OFF

The rest of parameters within this screen are not applicable to this communication mode with no server involved.

Press "Back" in order to go to "3. SIP Config", go to "4. Call Settings" and check that:

Media PORT is the one opened by the Network Administrator (paragraph 1.6, 5004 by default).

4. MAKING A TEST CALL TO PHOENIX MASTER

Finally, we are going to revise the adequate configuration of the equipment, and prepare it to make a test call to units called **phoenixMaster2** and **phoenixMaster**, that AEQ maintains for its customers, continuously transmitting audio.

Check that a set of headphones are connected to the front HP1 connection.

Main screen → Menu → ADVANCED → 2. Contacts. With the help of the user's manual, create a couple of new contacts with the following parameters:

Contact: **AEQMASTER_IP DIRSIP2**

Module Type: IP

URI: **phoenixMaster2@212.170.163.189**

Contact: **AEQMASTER_IP DIRSIP**

Module Type: IP

URI: **phoenixMaster@212.170.163.189:5061**

and make a call. In order to do so, from the main screen (Menu), select the "Select contact" line and choose:

Select contact: **AEQMASTER_IP DIRSIP2**

or

Select contact: **AEQMASTER_IP DIRSIP**

You can send the call by simply pressing the green key.

If the line "Select contact" presents a different name at its right, just select this line and press "Select" key. The call book will be presented. Just select **AEQMASTER_IP DIRSIP** or **AEQMASTER_IP DIRSIP2** from the list, press OK to close the list and then finally press the green key.

NOTE: Have in mind that this Call book entry that is created in the unit by default:

AEQMaster_IP

Module Type: IP

URI: **phoenixMaster@sip.aeq.es**

allows connection through AEQ SIP server only, with PROXY SIP option activated (ON). So it is not valid to connect to Phoenix Master test unit in this case (DIRECT SIP mode) and the above mentioned entries should be created.

- The top line at the screen displays (usually very fast) the status of the call, and the different phases it goes through: Calling, Synchronizing and finally **Connected.**
- The second line shows **AEQMASTER_IP DIRSIP2 o AEQMASTER_IP DIRSIP**
- The third line should show **G722** or the current audio coding scheme.

If the remote unit is busy, the top line will show "**Busy (486)**".

- Verify that the LED at the side of TX PROG SYNC key is lit **green**, as an indication that the communication has been successfully established and synchronized. If the key above HP1 is activated (illuminated) and the corresponding fader is open, the audio received from Phoenix Master should be heard at the headphones.

Once the connection is OK, you can retry as many times as needed varying the selected audio coding scheme in order to get the best possible quality allowed by the network (Wizard Menu → Sel Codec).

The list of available coding algorithms can be created in Menu → ADVANCED → 6. Link Profiles → Select

5. MAKING A CALL TO OTHER PHOENIX CODEC OR TO A N/ACIP COMPLIANT CODEC

The test call sent to Phoenix Master units has been automatically answered. In order to make a manual call to other codec, you must configure the second one the same way as the first unit, provided that it is also connected directly to a LAN with managed router. If it is not, just follow the corresponding Internet access application note.

Once the reporter equipment connects properly to Phoenix Master, you can create for convenience new entries in the call book: Menu → ADVANCED → 2. Contacts , with the name and URI of the remote unit, the same way it was done for Phoenix Master in the previous chapter, and then easily make calls to them.

You can of course also send the calls without having previously created a Call Book entry from the main screen, by pressing Menu to go to the Wizard. Select the Call to line, press Select and "URI" is selected. Then type the desired destination URI in the line below (select "Enter URI", then press OK and type the URI in).

A URI for DIRECT SIP use has the following format:

<codec_name>@<public_IP_of_the_codec> : <SIP port >

So, if the unit configured as in this example is called from another unit, they should dial as:

phoenix_123@212.170.180.170:5060

The call indications (optical and also acoustical, if enabled) will be activated at the destination codec, that will either wait for the user to answer the call or automatically off-hook (if this mode is enabled), and then establish the connection. The LED next to TX PROG should become green lit and audio should be heard through the headphones.

The three first screen lines should be:

Connected

<equipment name>@<remote IP_address>

<coding scheme>

For more information and other configuration options, please read the user's manual and Application Notes corresponding to other scenarios. If you still have doubts, please don’t hesitate to contact one of our assistance services:

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support@aeqbroadcast.com

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