

## Case study configuration Asterisk v. 1.4.18.1

On this version of Asterisk it is necessary to change constant in source code and then translate. This step might be solved all ready in any further versions. In version 1.6 should be video support much better. Asterisk in the case is not behaving in standard way however it works with it.

1. Change payload type for H.264 in Asterisk (Helios IP is using payload type no. 126 for video (H.264), Asterisk is using for H.264 no. 99 by default. There is need to change the H.264 payload type on Asterisk):

1. In the source code files, find out the RTP.c file (in the "main" folder). Find out this line "[99] = {1, AST\_FORMAT\_H264}," and modify the [99] to [126]. The result should look like

this:

```
static struct rtpPayloadType static_RTP_PT[MAX_RTP_PT] = {
.   [0] = {1, AST_FORMAT_ULAW},
.   #ifdef USE_DEPRECATED_G726
.   [2] = {1, AST_FORMAT_G726}, /* Technically this is G.721, but if Cisco can do it, so can we... */
.   #endif
.   [3] = {1, AST_FORMAT_GSM},
.   [4] = {1, AST_FORMAT_G723_1},
.   [5] = {1, AST_FORMAT_ADPCM}, /* 8 kHz */
.   [6] = {1, AST_FORMAT_ADPCM}, /* 16 kHz */
.   [7] = {1, AST_FORMAT_LPC10},
.   [8] = {1, AST_FORMAT_ALAW},
.   [9] = {1, AST_FORMAT_G722},
.   [10] = {1, AST_FORMAT_SLINEAR}, /* 2 channels */
.   [11] = {1, AST_FORMAT_SLINEAR}, /* 1 channel */
.   [13] = {0, AST_RTP_CN},
.   [16] = {1, AST_FORMAT_ADPCM}, /* 11.025 kHz */
.   [17] = {1, AST_FORMAT_ADPCM}, /* 22.050 kHz */
.   [18] = {1, AST_FORMAT_G729A},
.   [19] = {0, AST_RTP_CN},... /* Also used for CN */
.   [26] = {1, AST_FORMAT_JPEG},
.   [31] = {1, AST_FORMAT_H261},
.   [34] = {1, AST_FORMAT_H263},
.   [103] = {1, AST_FORMAT_H263_PLUS},
.   [97] = {1, AST_FORMAT_ILBC},
.   [101] = {0, AST_RTP_DTMF},
.   [110] = {1, AST_FORMAT_SPEEX},
.   [111] = {1, AST_FORMAT_G726},
.   [112] = {1, AST_FORMAT_G726_AAL2},
.   [121] = {0, AST_RTP_CISCO_DTMF}, /* Must be type 121 */
.   [126] = {1, AST_FORMAT_H264},
};
```

2. compile and install Asterisk with the previous change
2. edit the sip.conf file:
  1. insert this line in to the general section: videosupport=yes
  2. create the device (the simplest case, without authentication; the Helios IP has the phone number 1000):

```
[1000]
type=friend
context=phones
host=dynamic
allow=alaw
allow=ulaw
allow=h264
```

3. Insert the 1000 extension into the "phones" section in the extension.conf file:

exten => 1000,1,Verbose(1 | Helios IP)  
exten => 1000,n,Dial(SIP/1000,30)  
exten => 1000,n,Hangup()

4. Start (or reload) the Asterisk
4. set up the Helios IP:

The screenshot shows the HeliosIP SIP Settings web interface. The page title is "HeliosIP" and "SIP Settings". The interface is in English (EN). The left sidebar contains a navigation menu with categories: Information, Basic Settings, and Advanced Settings. Under Advanced Settings, the following options are listed: Network, Date and Time, SIP Settings (selected), Web Server, Audio, Video, Audio Codecs, Video Codecs, and Miscellaneous. Below the menu is a "Tools" section and a "Logout" button. The main content area is divided into three sections: "User settings", "SIP proxy settings", and "Registration at SIP proxy".

Section	Field	Value
User settings	Display name:	HeliosIP
	User ID:	1000
	Domain:	asterisk.server.com
	Use auth ID:	No
	Auth ID:	
SIP proxy settings	Proxy address:	asterisk.server.com
	Proxy port:	5060
	Use outbound proxy:	No
	OB proxy address:	
Registration at SIP proxy	Register Helios IP:	Yes
	Registration expires:	120 s
Other settings	Local SIP port:	5060

4. Save the configuration for Helios IP
5. On the Information page of Helios IP, there should be "Registered" in the Registration state line.