FAQ for Swissvoice IP 10

V1.1 – 12/07/2006

Upgrading

How to upgrade the software of an IP 10

To upgrade the firmware of IP 10, put the content of this zip file on a FTP or TFTP server. From the phone web page (log on as administrator: login=admin and password=admin), configure the file server link to provide the IP 10 with the FTP and/or TFTP configuration. Fill "swupdate_ip10.inf" into the "information file name" field, submit these parameters and then reboot the phone. During the boot process, IP 10 connects to the FTP or TFTP server to download swupdate_ip10.inf. In following it repeats the procedure for the new boot and application.

How to change the protocol on IP 10

To toggle from one protocol to another (MGCP to SIP for example), inside the swupdate_ip10.inf file (contained in the zipped file) it is mandatory to add the "update=forced" and "clean-data=all" options inside the **<APPLICATION>** tag, and the "update=forced" option inside the **<BOOT>** tag.

For example : <APPLICATION server=ftp update=forced clean-data=all>.

To upgrade the firmware of IP 10, put the content of this zip file on a FTP or TFTP server. From the phone web page (log on as administrator: login=admin and password=admin), configure the file server link to provide the IP 10 with the FTP and/or TFTP configuration. Fill "swupdate_ip10.inf" into the "information file name" field, submit these parameters and then reboot the phone. During the boot process, IP 10 connects to the FTP or TFTP server to download swupdate_ip10.inf. In following it repeats the procedure for the new boot and application.

How to configure several IP 10 at the same time?

The easiest way to have "plug and play" mechanism is to use a DHCP server to provide the IP 10 with option 66 (TFTP server address) and option 67 (file name to download "swupdate_ip10.inf").
When booting, the phone will request the swupdate_ip10.inf file from the TFTP server.
If you do not want to use option 66, you can provide FTP parameters with options 155, 156 and 157. In this case, the phone is downloading swupdate_ip10.inf using FTP information.
Once the swupdate_ip10.inf file has been downloaded, the IP 10 analyses the file for further procedure. Note that when filename is configured to "swupdate_ip10.inf", the phone tries first to download "macaddress_ip10.inf" in case there is a specific configuration for one or several phones.
Inside the <**CONFIG**> tag, you can add the configuration file to download. If you put "MACADDR_ip10.cfg", it looks for the file with the Mac address of the phone, for example 000590020101_ip10.cfg.

How to upgrade the software without deleting the phone book?

When upgrading the phone, the phone book is deleted entirely. To preserve the phone book, you need to save it first, then to download it back to the IP 10 after the upgrade.
Depending on the firmware release installed on the phone, you have 2 different solutions; the easier one does not apply to older firmwares.

For both solutions, connect as administrator on the IP 10 web interface.
Go to the "profiles" link and press the "save" button. Make sure that the common phone book section is shown in the displayed result. If yes, just copy this section into your cfg file (configuration file downloaded by the phone on start).

If you want to save the private phone book as well, you can also do it through the web interface, but you have to do it for each phone individually (as each phone has its own private phone book). Through the web, go to the "Import/export private phone book" link. Press the export button and save the content into a file (the format is CSV, as in the example below):

```
Name;Number
test1;2558852
test2;111111111
```

After the phone has been upgraded, the user can reload this file into the phone, using the "import" instead of the "export" button.

The same procedure can also be executed by the administrator for the common phone book. If the first way of getting the phone book is not working, proceed as for the private phone book. To avoid repeating this action on each phone, get the csv file and rewrite it in order to include it in the configuration file as follows:

```
#######################################
# PHONE BOOK parameters START
#######################################
set telbook CommonTelBookSource Local
set telbook del internal
set telbook add internal "test1" "2558852"
set telbook add internal "test2" "111111111"

#######################################
# PHONE BOOK parameters END
#######################################
```

How to use DHCP private option?

We attributed default affectation to some DHCP options, but you can change it through option 200 as described below.
The phone manages private options going from 150 to 159 and option 200.

Default affectation of the private option parameters:

```
150 - CA : Ip adress of the MGCP control server (Call Agent).
151 - GK : Ip adress of the H323 control server (Gatekeeper).
155 - FTP : Ip adress or Server-Name of the FTP server.
156 - FTPLog : Login of the FTP server.
157 - FTPPas : Password of the FTP server.
```

Please note: If option 200 is used, the default affectation is not defined.

It is possible to affect the default parameters to another id going from 150 to 159. To define them, you must use the option 200 and declare it as follows:

```
200 - "FTP=151; FTPLog=152;FTPPas=153;CA=150;"
Defines:
151 - FTP : Ip adress or Server-Name of the FTP server.
152 - FTPLog : Login of the FTP server.
153 - FTPPas : Password of the FTP server.
150 - CA : Ip adress of the MGCP control server (Call Agent).
```

What is the maximal length for pathname/filename for FTP/TFTP download?

The filename length is limited to 34 characters and the complete path with name is limited to 255 characters.

Phone book

IP 10 phone book specifications

There are 2 different phone books in IP 10: Private and Common.
The first one can only be local and uses csv format (it can be saved and uploaded by user).

Example:

Name;Number
Dupont;104450
Durant paul;056684

The common phone book can be local or remote. When it is local, it also uses csv format and can be uploaded/saved by admin.
You can also add contacts with the config.sys, which is uploaded during the boot sequence, using the following command:
```
set telbook add internal/private "<Name>" "<Number>"
```
internal/private allows you to choose between the internal (common) or private phone book.
If «name» or «number» length are too long, the string is truncated to the maximum length (24 for name and 26 for number).

Limitation for these phone books:
Maximum characters for Name = 24
Maximum characters for Number = 26
Private phone book = up to 100 entries.
Local common phone = up to 400 entries.

Usage

Is it possible to use the phone's own number to call the voicemail service?

Example: is it possible to call 123 to access the voicemail when the phone extension is 123?

It is possible on SIP firmware from version v1.0.1b2. This access mode can only be used to call the voicemail and the service has to support it. This feature is not relevant on MGCP and H323.

Can an IP 10 SIP directly connect to another IP 10 (peer-to-peer)?

IP 10 SIP supports peer-to-peer calls. In that case you need to modify the "atpm table" (numbering table).

In following example, these commands need to be entered through Telnet, or you can include them inside a configuration file:

```
##################################################
# Build ATPM peer to peer config for IP10 phones SIP#
##################################################
atpm req    # do not add this command inside a CFG file
atpm aadd 100 3 3 80 3
atpm dadd 80 rvcm 100@149.1.10.100

atpm aadd 101 3 3 81 3
atpm dadd 81 rvcm 101@149.1.10.101

atpm aadd 102 3 3 82 3
atpm dadd 82 rvcm 102@149.1.10.102

atpm aadd 300 3 3 83 3
atpm dadd 83 rvcm 300@149.1.10.1

atpm aadd 301 3 3 84 3
atpm dadd 84 rvcm 301@149.1.10.2

atpm genlist hunt
atpm done    # do not add this command inside a CFG file
atpm store   # do not add this command inside a CFG file
##################################################
```

You have to register each phone you want to call. This is done with two commands:

First command: `atpm aadd 100 3 3 80 3`
This means to send the call with extension number 100 to destination number 80. Minimum number is 3, maximum size is 3, and it means that the number is complete.
The exact syntax of this command is:

```
aadd <tele_number> <min_digits> <max_digits> <hunt_group_id>
<prefix_strip_len> [prefix_number]
```

Second command: `atpm dadd 80 rvcm` 100@149.1.10.100
Destination 80 corresponds to SIP address: 100@149.1.10.100
The exact syntax of this command is:
```
dadd    <dest_id> <rvcm> <dest related params>
```


Can an IP 10 H323 directly connect to another IP 10 (peer-to-peer)?

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In following an example. These commands need to be entered through Telnet, or you can include them inside a configuration file:

```
###################################################
# Build ATPM peer to peer config for IP 10 MGCP      #
###################################################
atpm req   # do not add this command inside a CFG file
atpm aadd 100 3 3 80 0
atpm dadd 80 rvcm TA:149.1.10.100,NAME:TOPTEL,TEL:100

atpm aadd 101 3 3 81 0
atpm dadd 81 rvcm TA:149.1.10.101,TEL:101

atpm aadd 102 3 3 82 0
atpm dadd 82 rvcm TA:149.1.10.102,TEL:102

atpm aadd 300 3 3 83 0
atpm dadd 83 rvcm 300@149.1.10.1
atpm dadd 83 rvcm TA:149.1.10.1,TEL:300

atpm aadd 301 3 3 84 0
atpm dadd 84 rvcm TA:149.1.10.2,TEL:301

atpm genlist hunt
atpm done    # do not add this command inside a CFG file
atpm store   # do not add this command inside a CFG file
###################################################
```
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<prefix_strip_len> [prefix_number]
```

Second command: `atpm dadd 80 rvcm TA:149.1.10.100,NAME:TOPTEL,TEL:100`
Destination 80 correspond to: 149.1.10.100

The exact syntax of this command is:
```
dadd   <dest_id> <rvcm> <dest related params>
```

<dest related params> are:
TA:DESTINATION_IP_ADRESS
NAME:DESTINATION_INFO_NAME (optionnal)
TEL:DESTINATION_PHONE_NUMBER (optionnal)

Does IP 10 support the conference feature?

IP 10 with SIP protocol is not handling conference as an internal feature. You need to have access to a MCU (Multipoint Conferencing Unit) to do this. This feature may be included in your proxy server.

What is the maximum length for the SIP user ID?

Maximum length is 29 characters.

How to disable the call waiting feature on IP 10?

It is not possible with H323 but it is with MGCP (configured on the call manager) and for SIP from release v1.0.0b19.
To disable call waiting with SIP, proceed as follows in the phone interface:
Menu (key ok) -> User Settings -> Other Settings -> Call Waiting -> On/Off

Network

Does IP 10 support NAT?

There is no NAT support for IP 10 with H323.
With SIP firmware, NAT support is limited to public network address definition.
There is no STUN support on IP 10.

How to define the base RTP port of the IP 10?

IP 10 uses RTP port 30 000 as base (and the next 3 ports). But you can use the full range of ports; you just have to be careful not to choose a port already taken by another defined protocol.
You can change the RTP port with the following sequence:

Log on to the Telnet interface (login: target/ password: password)

Type dbgcmd to get the MXP> prompt.
```
MXP> set xgcp RtpBasePort <Port value [1024..65535]>
```

Continue as follows:
```
MXP> activate
MXP> commit
```
to save the parameters.

DTMF

How to check DTMF behaviour (in-band/out-of-band) with MGCP?

The easiest way to check the DTMF settings is to use the phone web interface. Log on as administrator (admin/admin), go to the Profiles link and click on Save.
Your current phone profile will be displayed. Here you can look for the following lines:
```
set coding 1 dtmf_relay on oob
set tcid 0 ipp kpad_echo on
set utilities patches dtmf_voice off
set utilities patches dtmf_RFC2833 off
```
Refer to the Telnet commands guide Valid_21-dec-05_IP 10.pdf to get more information about these commands.

What to do if some DTMF are missing?

If analysing ethereal capture, you see that some of the DTMF are missing; you need to check that `kpad_echo` is set to off.
Connect to the Telnet interface of IP 10 (login: target, password: password), then enter **dbgcmd** to get the MXP> prompt. Check the value with this command:
```
MXP>show tcid 0
```
If the value is not correct, use this command:
```
set tcid 0 ipp kpad_echo off
```

TONE TABLE

How to configure numbering tables?

If you enter the "atpm alist" command in a Telnet session, you will get:

```
MXP>atpm alist
                Address Hunt   Min    Max    Prefx Prefix
                Entry   Grp_Id Digits Digits strip Address

                   0     10     1     26       0   None
                   1     10     1     26       0   None
                   2     10     1     26       0   None
                   3     10     1     26       0   None
                   4     10     1     26       0   None
                   5     10     1     26       0   None
                   6     10     1     26       0   None
                   7     10     1     26       0   None
                   8     10     1     26       0   None
                   9     10     1     26       0   None
                   *     10     1     26       0   None
                   #     10     1     26       0   None
OK
```

If you want to manage more precise prefixes and numbers, here an example:
In our default tables we allow all numbers from 1 to 26 digits length. If for example you want to say that numbers beginning with 456 are 8 digits long and other numbers beginning with 4 are 5 digits long, you have to proceed as follows under Telnet (or in a configuration file):

atpm req   ==> this request opens our numbering database
atpm purge addr   ==> this request deletes all registered prefixes (0,1,2,3,4,5,6,7,8,9,*,# in our case)
atpm purge hunt   ==> deletes internal "hunt group"

atpm aadd 456 8 8 10 0   ==> we create prefix 456, minimum length is 8, maximum length is 8, associated destination is 10 (this destination is linked to the proxy server ip address, so it is mandatory to keep this value)
atpm aadd 4 5 5 10 0   ==> we create prefix 4, minimum length is 5, maximum length is 5, associated destination is 10, and we do not strip any digits

atpm genlist hunt   ==> we generate internal "hunt group"
atpm done   ==> we close the database
atpm store   ==> we store the database into the flash memory.

Then all other prefix definitions you need to enter have to be inserted with "atpm aadd list".

How to add a prefix to a number using the numbering table?

You can tell the numbering table to add digits 91 in front of each phone number beginning with something different than "91".

If you enter the "atpm alist" command in a Telnet session, you will get:

```
MXP>atpm alist
                   Address Hunt   Min     Max     Prefx Prefix
                   Entry   Grp_Id Digits Digits  strip Address

                      0     10     1      26       0    None
                      1     10     1      26       0    None
                      2     10     1      26       0    None
                      3     10     1      26       0    None
                      4     10     1      26       0    None
                      5     10     1      26       0    None
                      6     10     1      26       0    None
                      7     10     1      26       0    None
                      8     10     1      26       0    None
                      9     10     1      26       0    None
                      *     10     1      26       0    None
                      #     10     1      26       0    None
OK
```

It needs to be modified to get the following result:

```
                   Entry   Grp_Id Digits Digits strip Address

                      0     10     1      26       0    91
                      1     10     1      26       0    91
                      2     10     1      26       0    91
                      3     10     1      26       0    91
                      4     10     1      26       0    91
                      5     10     1      26       0    91
                      6     10     1      26       0    91
                      7     10     1      26       0    91
                      8     10     1      26       0    91
                      9     10     1      26       0    91
                     91     10     2      26       0    None
                      *     10     1      26       0    None
                      #     10     1      26       0    None
```

To do this, inside configuration file, proceed as follows:

```
atpm req
atpm purge addr
atpm purge hunt
atpm aadd 0 1 26 10 0 91
atpm aadd 1 1 26 10 0 91
atpm aadd 2 1 26 10 0 91
atpm aadd 3 1 26 10 0 91
atpm aadd 4 1 26 10 0 91
atpm aadd 5 1 26 10 0 91
```

```
atpm aadd 6 1 26 10 0 91
atpm aadd 7 1 26 10 0 91
atpm aadd 8 1 26 10 0 91
atpm aadd 9 1 26 10 0 91
atpm aadd 91 2 26 10 0
atpm genlist hunt
atpm done
atpm store
```

It means that all numbers dialled on the phone will be prefixed with "91". (Even the numbers dialled manually).
If you want to be able to make internal calls without prefixing numbers with "91", you need to specify in this table for which prefixes you don't want to add the "91" prefix (it is the case for numbers beginning with 91).
For example if your internal numbers begin with 32, you need to insert the following line:

```
atpm aadd 32 2 26 10 0
```

What is the tone table for Europe?

The tone table definition for Europe is:

```
set tone nbElem BZ 2
set tone element BZ 1 500 425 -60
set tone element BZ 2 500 0 0

set tone nbElem CG 2
set tone element CG 1 200 425 -60
set tone element CG 2 200 0 0

set tone nbElem DL 1
set tone element DL 1 inf 425 -60

set tone nbElem DT 4
set tone element DT 1 250 480 -240 620 -240
set tone element DT 2 200 0 0
set tone element DT 3 250 480 -240 620 -240
set tone element DT 4 inf 0 0

set tone nbElem BDL 3
set tone element BDL 1 150 523 -130
set tone element BDL 2 150 784 -130
set tone element BDL 3 2000 0 0

set tone nbElem OT 2
set tone element OT 1 500 425-60
set tone element OT 2 500 0 0

set tone nbElem RO 2
set tone element RO 1 250 480 -240 620 -240
set tone element RO 2 250 0 0

set tone nbElem RT 2
```

```
set tone element RT 1 1000 425 -60
set tone element RT 2 4000 0 0

set tone nbElem WT1 2
set tone element WT1 1 300 440 -130
set tone element WT1 2 inf 0 0

set tone nbElem WT2 4
set tone element WT2 1 100 440 -130
set tone element WT2 2 100 0 0
set tone element WT2 3 100 440 -130
set tone element WT2 4 1700 0 0

set tone nbElem WT3 6
set tone element WT3 1 100 440 -130
set tone element WT3 2 100 0 0
set tone element WT3 3 100 440 -130
set tone element WT3 4 100 0 0
set tone element WT3 5 100 440 -130
set tone element WT3 6 1500 0 0

set tone nbElem WT4 6
set tone element WT4 1 100 440 -130
set tone element WT4 2 100 0 0
set tone element WT4 3 300 440 -130
set tone element WT4 4 100 0 0
set tone element WT4 5 100 440 -130
set tone element WT4 6 1300 0 0
```

Hardware

How works the IP 10 internal switch?

Internal switch is 10/100 Mbps auto sense and auto MDX (recognize straight and cross Ethernet cables).

The phone does not modify nor tag the data flow coming from or going to the PC. From PC side, the phone is totally transparent. Other equipments, if they are configured to use another VLAN, have no problem to interact with the PC. The internal switch of the IP 10 is not a software function of the phone; it is a separate component inside the phone (basically explained). The software of the IP 10 never modifies or analyses data coming from/going to the PC as it never gets this data. In some way, the internal switch acts more as a hub. If the LAN incoming flow is not for the phone, the switch transfers it to the PC port regardless the PC is the final destination or not.

What is the IP 10 power required to use PoE?

IP 10 follows the IEEE 802.3af (http://www.ieee802.org/3/af/requirements.pdf) requirement for a nominal voltage of 48V.

At boot time, the in-rush current at PoE input stays lower than 400 mA (it can reach this value for only less than 50 ms). During a call, the IP10 uses 100 mA, otherwise only 60 mA.

Power consumption is 4.8 W in use and 2.9 W in "stand by" (no call).

It supports powering over the non data wires and via phantom supply using the data wires.

IP 10 does not support classification, it is considered as class 0 by a power switch.

If PoE is connected first, the phone starts with the PoE and uses it. If PoE fails, the phone switches immediately on power supply without interruption.

If Power Supply is connected first, PoE is not detected. If Power Supply fails, the phone stops, then PoE is detected and IP 10 boots.

In the first case, transition is not visible, in the second a reboot is done.

Which headsets are compatible with IP 10?

You can check in document: Headsets Compatibility List.pdf

Otherwise, every headset with the following specifications should work with IP 10:

- RJ11 connector with microphone on pins 1 - 4 (+ on pin 4) and 2 - 3 for speaker.

-150 ohms as nominal impedance for speaker.

- Electrets condenser microphone with a 4mA maximal current of polarization.

Audio

How to adjust echo in handset mode for SIP?

To see the current handset mode audio profile, enter the following command (the values to be changed are displayed in red):

```
MXP>show ipp_gains 0
Gain Table for Profile 0
index   rx_dg(1 dB) rx_ag rx_sec_ag aer_rx_dg aer_nlp_clip(3 dB) aer_nlp_clt
=============================================================================
    0  0           0      -48       0         0                  6000
    1  0           0      -32       0         0                  5000
==> 2  0           0      -25       0         0                  4000
    3  0           0      -17       0         0                  3000
    4  0           0      -10       0         0                  2000


and command:

MXP>show ipp_prof 0

Profile 0
=========
usage_bitmap:          0x1
                       handset
aer_config:            0x1
                       AER_DISABLE
                       AER_UPDATE
                       NLP_ENABLE
                       AER_ADAPTIVE_ON_MUTE_DISABLE
                       HALF_DUPLEX_DISABLE
                       AER_TX_IDLE_ENABLE
aer_tail:              10
aer_tx_dg (0.5 dB):    0
aer_min_rx_noise_thresh: 5380
aer_min_tx_noise_thresh: 5380
aer_rx_to_tx_hangover: 30
aer_tx_to_rx_hangover: 10
aer_rx_linear_threshold: 24576
agc_control:           0x0
                       AGC_DISABLE_OFF
                       AGC_ADAPTIVE_OFF
sat_thresh:            32500
sat_hangover:          20
tx_ag: (0.5 dB):       67
tx_dg: (1 dB):         0
Side Tone Gains:(0.5 dB):10
noise_guard:           off
hangover_period:       default
send_noise_level:      default
ramping_in_period:     default
input_loss:            default
rx_ag(min,max,default)  0, 0, 0 ( 0.5 dB)
```

Then you need to have (new values in red):

```
MXP>show ipp_gains 0
Gain Table for Profile 0

Index rx_dg(1 dB) rx_ag rx_sec_ag aer_rx_dg aer_nlp_clip(3 dB) aer_nlp_clt
=========================================================================
    0 0          0     -48       0         0                  32767
    1 0          0     -32       0         0                  32767
==> 2 0          0     -25       0         0                  20000
    3 0          0     -17       0         0                  15000
    4 0          0     -10       0         0                  9000
```

and command:

```
MXP>show ipp_prof 0

Profile 0
=========
usage_bitmap:          0x1
                       handset
aer_config:            0x1
                       AER_ENABLE
                       AER_UPDATE
                       NLP_ENABLE
                       AER_ADAPTIVE_ON_MUTE_DISABLE
                       HALF_DUPLEX_ENABLE
                       AER_TX_IDLE_ENABLE
aer_tail:              10
aer_tx_dg (0.5 dB):    0
aer_min_rx_noise_thresh: 5380
aer_min_tx_noise_thresh: 5380
aer_rx_to_tx_hangover: 30
aer_tx_to_rx_hangover: 10
aer_rx_linear_threshold: 24576
agc_control:           0x0
                       AGC_DISABLE_OFF
                       AGC_ADAPTIVE_OFF
sat_thresh:            32500
sat_hangover:          20
tx_ag: (0.5 dB):       67
tx_dg: (1 dB):         0
Side Tone Gains:(0.5 dB):10
noise_guard:           off
hangover_period:       default
send_noise_level:      default
ramping_in_period:     default
input_loss:            default
rx_ag(min,max,default) 0, 0, 0 ( 0.5 dB)
```

To enter these new values, use the following commands under the MXP> prompt:

```
set ipp_gains 0 aer_nlp_clt 0 32767
set ipp_gains 0 aer_nlp_clt 1 32767
```

```
set ipp_gains 0 aer_nlp_clt 2 20000
set ipp_gains 0 aer_nlp_clt 3 15000
set ipp_gains 0 aer_nlp_clt 4 9000

set ipp_prof 0 aer on
set ipp_prof 0 aer_half_duplex enable

activate
commit
```

This new configuration for handset mode activates echo canceller for the 3 highest volumes in this mode (no impact if you enter value 32767 for the 2 lowest values). If you want to adjust these values depending on your tests, you have to decrease the aer_nlp_clt value which will reduce the echo, but will increase half duplex.

The same adjustment can be done for handsfree mode (profile 2) and group listening mode (profile 3).

How to check and modify acoustic adjustments for SIP?

Find below a description of the commands which show the acoustic adjustments and how you can change them.

```
MXP>show ipp_prof all.
```

Inside these parameters emission can be changed with "set ipp_prof ...".

6 profiles are displayed (0 to 5). Do not consider profile 5.
Then:

```
MXP>show ipp_gains all
```

You get the configuration for all audio profiles.
The configuration can be modified with "set ipp_gains ...".

Microphone sensitivity can be adjusted for each profile using commands:
set ipp_prof <prof> tx_ag <gain> (in 0.5 dB steps) set ipp_prof <prof> tx_dg <gain> (in 1 dB steps) Ag is anolog gain, and dg is digital gain.

We do not recommend to modify these parameters because many parameters have an influence on other parameters and it is really difficult to find the right adjustments for all situations.

Other values that you can try to change for handset profile (profile 0) are the following: "Aer_nlp_clip" value and "aer_nlp_clt" value, which can be between 0 and 32768. The lower the value of aer_nlp_clt is, the better quality of full duplex you have, but less echo suppression. In order that these values have an effect, you have to activate echo canceller for this profile, using the "set ipp_prof 0 aer on" command.

The aer_nlp_clip value, initiated at -10, is increased until echo reduction is acceptable. This one is especially conservative against using a high value for aer_nlp_clip, as customers expect full duplex performance from a handset.

How to adjust echo with MGCP?

To reduce the echo on IP 10, you can activate the following patch from the Telnet interface:
```
set utilities patches handset_gains [on|off]
```

You can check whether the patch is correctly set with the command:
```
show utilities patches
```

How to select where the phone rings (headset or speaker)?

```
set utilities headset_ring_mode [LOUDSPEAKER|HEADSET]
```

To have a short ring, you can create an RTTTL melody, or you can use the set of commands below:

```
MXP>set ringtones
COMMANDS TO SET RING TONES :
  set ringtones RTTTL <Id[1..5]> <source[FTP|TFTP]> <filename>
  set ringtones RTTTL delete <all|[1..5]>
  set  ringtones  cadence  <tone[R0..R7|RS|RG]>  <sequence[1..4]>  <ON(ms)>
<OFF(ms)>
  set ringtones cadence DEFAULT <all|tone[R0..R7|RS|RG]>
  set  ringtones  melodymap  <tone[R0..R7|RS|RG]>  <DEFAULT|melodyId[1..15]>
<adjust duration[TRUE|FALSE]
  set ringtones melodymap DEFAULT <all|tone[R0..R7|RS|RG]>
  set ringtones name <tone[R0..R7|RS|RG]> <DEFAULT|NOT_USED|name>
MXP>
```

To disable tones, you first need to activate "configurable tone table" with command:
set utilities tone_table CONF, then you need to redefine each tone you want to "suppress", using the following set of commands:

```
MXP>set tone
COMMANDS TO SET TONE TABLE ELEMENTS :
  set tone nbElem ToneId <nb of elements[1 to 6]>
  set tone element ToneId <elementID> <duration in ms> <F1> <A1> [<F2> <A2>
<F3> <A3> <F4> <A4>]
  set tone default
  show tone ToneId
```

With ToneId:
  -> BZ.............Busy tone
  -> CG.............Congestion tone

```
-> DL.............Dial tone
-> BDL..........Breaking dial tone
-> DT.............Disconnection tone
-> OT.............Off Hook Notice tone
-> RO.............Reorder tone
-> RT.............Ring Back tone
-> SL.............Stutter Dial tone
-> WT1.............Call waiting 1 tone
-> WT2.............Call waiting 2 tone
-> WT3.............Call waiting 3 tone
-> WT4.............Call waiting 4 tone  Some tones are linked:
-> Confirmation tone...............BDL (600 ms)
-> Message Waiting Ind tone........BDL (infinite)
```

For example, to have a silent dial tone, use commands:

```
set tone nbElem DL 1    /* this command says that DL is composed of 1 tone
*/
```

set tone element DL 1 0 0 0 /* then configure tone "1" of DL(dial tone) and play a frequency which is 0 */

Should you have further questions, please feel free to send us an e-mail to:
support_ip@swissvoice.net