How to record conversations in IP0x

ATCOM®

Product Guide

Version: 2.0

2012-11-27

Content

Contact ATCOM

The Introduction of ATCOM

Founded in 1998, ATCOM technology has been always endeavoring in the R&D and manufacturing of the internet communication terminals. The product line of ATCOM includes IP Phone, USB Phone, IP PBX, VoIP gateway and Asterisk card.

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How to record conversations in IP0x

This function is achieved by adding MixMonitor() application in Asterisk dial plan (/persistent/etc/asterisk/extensions.conf). please open the file via SSH or FTP.

SSH (user/password: root/12xerXes16)

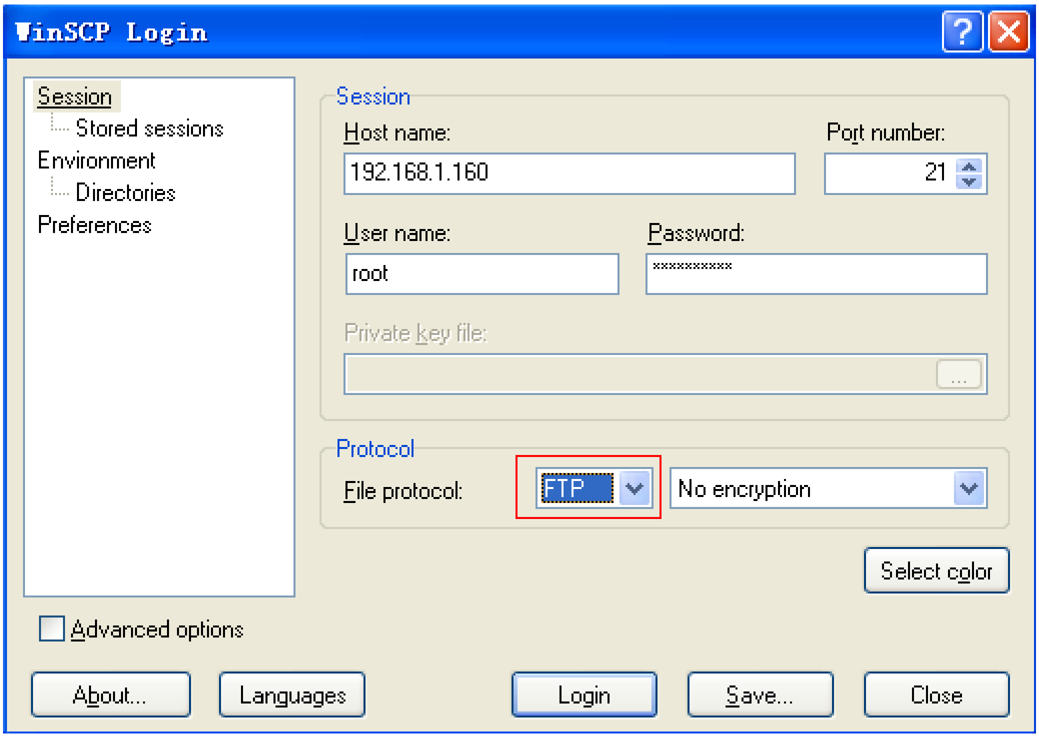
If you are familiar to Linux command, you can edit it via SSH.

Putty download address: http://download.atcom.cn:8080/IPPBX/putty.exe

FTP(user/password: root/12xerXes16)

If you want to edit it in Windows, you can use WinSCP.

Download address: http://winscp.net/download/winscp437.zip



1. Record interior calls and inbound calls

[macro-stdexten] code segment is charge of the calls to extensions. The default settings should be like below:

```
[macro-stdexten]
exten => s,1,Dial(${ARG2},200,tTkK)
exten => s,n,NoOp(${QUEUEFLAG})
exten => s,n,GoToIf($["${QUEUEFLAG}" != ""]?queues,${QUEUEFLAG},1:s-${DIALSTATUS},1)
exten => s-NOANSWER,1,GoToIf($[$["${FOLLOWME_${ARG1}}" = "1"] & $["${follow}" != "7" ]]?followme,${ARG1},1:s-NOANSWER,2)
exten => s-NOANSWER,2,Voicemail(${ARG1},u)
exten => s-NOANSWER,n,Goto(default,s,1)
exten => s-BUSY,1,Voicemail(${ARG1},b)
exten => s-BUSY,2,Goto(default,s,1)
exten => _s-.,1,Goto(s-NOANSWER,1)
exten => a,1,VoicemailMain(${ARG1})

- extensions.conf 90/299 30%
```

1）Please change it to as below:

```
[macro-stdexten]
exten=>s,1,MixMonitor(/persistent/sounds/record/${ARG1}-${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)}.wav,ab)
exten => s,2,Dial(${ARG2},60)
exten => s,3,StopMonitor()
exten => s,4,Goto(s-${DIALSTATUS},1)
exten => s-NOANSWER,1,GoToIf($[$["${FOLLOWME_${ARG1}}" = "1"] & $["${follow}" != "7" ]]?followme,${ARG1},1:s-NOANSWER,2)
exten => s-NOANSWER,2,Voicemail(${ARG1},u)
exten => s-NOANSWER,3,Goto(default,s,1)
exten => s-BUSY,1,Voicemail(${ARG1},b)
exten => s-BUSY,2,Goto(default,s,1)
exten => _s-.,1,Goto(s-NOANSWER,1)
exten => a,1,VoicemailMain(${ARG1})
```

/persistent/sounds/record/ is the directory for storing recordings, it can be the SD card (change it to **/mnt/sd/**) or NFS directory (see section 5).

${ARG1}-${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)}.wav is the name of the recordings:

${ARG1} is a variable standing for callee's number.

${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)} is a variable standing for recording time

.wav stand for the format of recordings, you also can use .gsm,.ulaw etc.

After someone make a call to 6001, below is the recording file:

```
root:/persistent/sounds/record> ls
6001-20120223065539.wav
```

Also you can set the name of the recordings to

${CALLERID(num)}-${ARG1}-${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)}.wav

${CALLERID(num)} is a variable standing for caller's number

2）Reload Asterisk to make the changes take effect

root:~> **asterisk -r**

Asterisk 1.4.21.2, Copyright (C) 1999 - 2008 Digium, Inc. and others.

Created by Mark Spencer <markster@digium.com>

Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.

This is free software, with components licensed under the GNU General Public

License version 2 and other licenses; you are welcome to redistribute it under

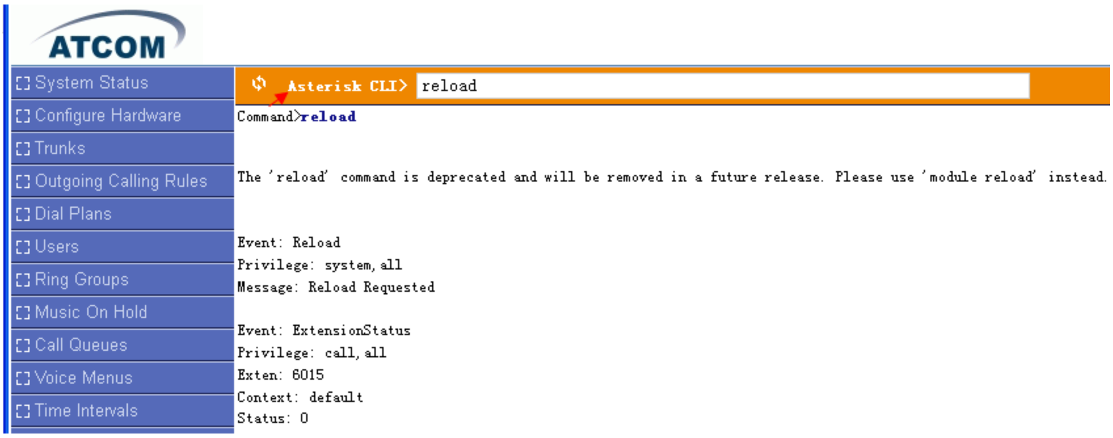
certain conditions. Type 'core show license' for details.

===

Connected to Asterisk 1.4.21.2 currently running on ip02 (pid = 174)

ip02*CLI> **reload**

Or



2. Record appointed conversation

If you want to record all calls to extension 6005, you can change the [macro-stdexten] code segment into below one:

```
[macro-stdexten]
exten => s,1,GotoIF($["${ARG1}"="6005"]?4:2)
exten => s,2,Dial(${ARG2},20)
exten => s,3,Goto(s-${DIALSTATUS},1)
exten=>s,4,MixMonitor(/persistent/sounds/record/6005-${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)}.gsm)
exten => s,5,Dial(${ARG2},20)
exten => s,6,StopMonitor()
exten => s,7,Goto(s-${DIALSTATUS},1)
exten => s-NOANSWER,1,GoToIf($[$["${FOLLOWME_${ARG1}}" = "1"] & $["${follow}" !=
"7" ]]?followme,${ARG1},1:s-NOANSWER,2)
exten => s-NOANSWER,2,Voicemail(${ARG1},u)
exten => s-NOANSWER,3,Goto(default,s,1)
exten => s-BUSY,1,Voicemail(${ARG1},b)
exten => s-BUSY,2,Goto(default,s,1)
exten => _s-.,1,Goto(s-NOANSWER,1)
exten => a,1,VoicemailMain(${ARG1})
```

3. Record Outbound calls

1) Find outgoing calling rule

Please create outgoing calling rule for the trunk via WEB GUI, and then find it in /etc/asterisk/extensions.conf. It is named as [CallingRule_trunkname]

```
[CallingRule_to159]
exten = _0X,1,Macro(trunkdial-failover-0.3,${6010}/60${EXTEN:0},,6010,)
```

2) Change it to below:

[CallingRule_to159]
exten =
_0X,1,MixMonitor(/persistent/sounds/record1/${EXTEN}-${STRFTIME(${EPOCH},,%C%y%m%d%H%M%S)}.wav,b)
exten = _0X,n,Macro(trunkdial-failover-0.3,${6010}/60${EXTEN:0},,6010,)
exten = _0X,n,StopMonitor()

3) Reload Asterisk

4. One touch recording

We can dial a feature code to begin recording and dial it again to end the recording over a conversion.

1) Set feature code for one touch recording in /etc/asterisk/features.conf

[featuremap]
automon = *0
[applicationmap]
automon=> *0,self/both,Macro,apprecord

2) Add below recording function in /etc/asterisk/extensions.conf

[macro-apprecord]
exten => s,1,GotoIf($["${XAD}"="0" | "${XAD}"=""]?startrec:stoprec)
exten => s,n(startrec),Playback(beep)
exten => s,n,Set(XAD=1)
exten =>
s,n,Set(FILENAME=${STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)}-${CALLERID(num)}-${ARG1}.WAV)
exten => s,n,MixMonitor(/persistent/sounds/record/${FILENAME},ab)
exten => s,n,MacroExit
exten => s,n(stoprec),StopMixMonitor
exten => s,n,Set(XAD=0)
exten => s,n,Playback(beep)
exten => s,n,MacroExit

3) Declare the feature code in interior call function [macro-stdexten]

[macro-stdexten]
exten => s,1,Set(__DYNAMIC_FEATURES=automon); add this extension is OK
exten => s,**n**,Dial(${ARG2},20,tTkK) ;the priority change to n from 1
exten => s,n,NoOp(${QUEUEFLAG})
exten => s,n,GoToIf($["${QUEUEFLAG}" != ""]?queues,${QUEUEFLAG},1:s-${DIALSTATUS},1)
exten => s-NOANSWER,1,GoToIf($[$["${FOLLOWME_${ARG1}}" = "1"] & $["${follow}" !=

"7"]]?followme,${ARG1},1:s-NOANSWER,2)

exten => s-NOANSWER,2,Voicemail(${ARG1},u)

exten => s-NOANSWER,n,Goto(default,s,1)

exten => s-BUSY,1,Voicemail(${ARG1},b)

exten => s-BUSY,2,Goto(default,s,1)

exten => _s-.,1,Goto(s-NOANSWER,1)

exten => a,1,VoicemailMain(${ARG1})

4) Reload Asterisk

5. How to mount NFS to IP0x

1) Check your NFS server

My server IP is 192.168.1.241 and the NFS directory is /home/bty/

```
[root@bty bty]# ifconfig
eth0      Link encap:Ethernet  HWaddr 00:0C:29:04:89:2A
          inet addr:192.168.1.241  Bcast:192.168.1.255  Mask:255.255.255.0
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:125692 errors:0 dropped:0 overruns:0 frame:0
          TX packets:2151 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:10808516 (10.3 MiB)  TX bytes:180461 (176.2 KiB)
          Interrupt:67 Base address:0x2000

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:2391 errors:0 dropped:0 overruns:0 frame:0
          TX packets:2391 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:4032466 (3.8 MiB)  TX bytes:4032466 (3.8 MiB)

[root@bty bty]# showmount -e
Export list for bty.com:
/home/bty *
```

2) Mount NFS directory to an empty directory in IP0x

root:/mnt/uba> **mount -o nolock -t nfs 192.168.1.241:/home/bty /mnt/uba**

root:/mnt/uba> **ls**

root:/mnt/uba> **df**

Filesystem	1k-blocks	Used	Available	Use%	Mounted on
/dev/mtdblock0	14327	14049	278	98%	/
/dev/mtdblock2	253952	134032	119920	53%	/persistent
192.168.1.241:/home/bty	4956320	209184	4491296	4%	/mnt/uba

```
root:/mnt/uba> cd /mnt/uba
root:/mnt/uba> ls
stream.raw              stream.raw.wav          hardware.html
vimrctest               Desktop                 streamcid.raw
streamcid.raw.wav       op_panel-0.30.tar.gz
op_panel-0.30           test
```

3) Make the mount permanent

However, the NFS directory will be unmounted once IPPBX reboot, we need to let the PBX mount the directory automatically when it boot. Adding a script in /etc/rc.d/ will solve the problem, the file named begin S in this directory will be executed when IPPBX boot.

```
root:/mnt/uba> cd /etc/rc.d/
root:/persistent/etc/rc.d> ls
S60IPtables   S50asterisk   S40zaptel     S99local
S35cron       S30ntp        S10network
root:/persistent/etc/rc.d> echo "mount -o nolock -t nfs 192.168.1.241:/home/bty /mnt/uba" > S70nfs
root:/persistent/etc/rc.d> cat S70nfs
mount -o nolock -t nfs 192.168.1.241:/home/bty /mnt/uba
root:/persistent/etc/rc.d> chmod +x S70nfs
```

Also you can mount the NFS to an not empty directory, like /persistent/sounds/record/ ,once the mount operation is successful , the original content will be hided. For example, if you mount the NFS to /persistent/sounds/voicemail/default/ , all the voicemail will be stored to your NFS server.

Other information:

If you want to know more about MixMonitor(), please refer to below link:

http://www.asteriskguru.com/tutorials/mixmonitor.html

http://www.asteriskguru.com/tutorials/monitor.html

and more information on feature codes:

http://www.voip-info.org/wiki/view/Asterisk+config+features.conf

$｛STRFTIME()｝

http://www.kernel.org/doc/man-pages/online/pages/man3/strftime.3.html