3CX + PBXMate Set up guide

After having various issues reported with call quality, I began looking for a solution that would fix, monitor or improve call quality problems especially an intermittent echo heard by the inbound caller.

We found the main cause to be related to the headsets used (USB headsets and a softphone) causing an echo to be heard by the external party, and also the amount of ambient noise in the office – people talking across the room, the coffee machine, agent typing on the keyboard ect…

Our options would be to investigate and test different headsets to find one that provides the noise and echo suppression, or a softphone that does this, or some other software run on the PBX monitoring the audio on the trunk line.

Enter PBXMate - software that could minimise this ambient noise and provide echo cancellation between 3CX and our SIP trunk service provider, thereby working with the existing headsets and hardware currently in use.

This is a quick look at how I set up to test, and some examples of the difference that the software makes to the call.

Please note the information contained in this write up is as applied to my own test environment, it is not 3cx or Solicall documentation and may or may not be applicable to your own environment.

It is intended as a reference only.

1: Overview

PBXMate will run on a separate machine, and I will make the changes required to implement PBXMate between 3cx and our VoIP Provider, like so:
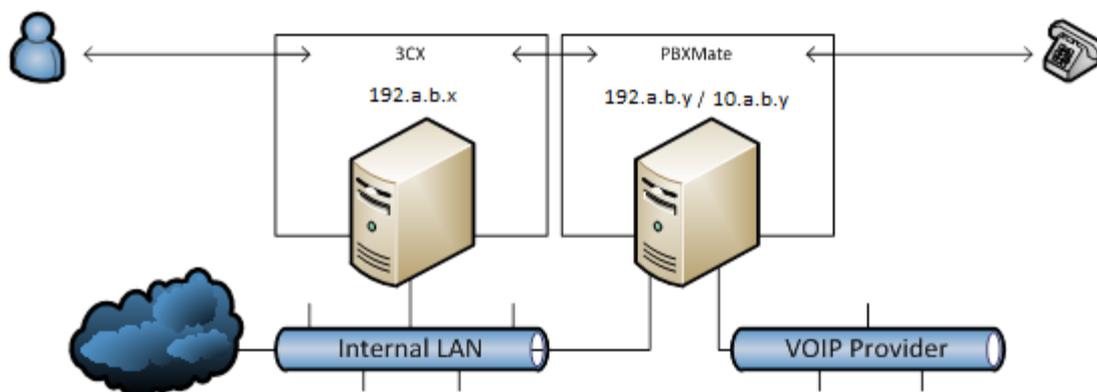
3CX (Windows SBS Server 2008)
LAN interface 1 (192.a.b.x /24) connects to the internal network
LAN interface 2 DISABLED (10.a.b.y /24) was connected to our VoIP providers network.

PBXMate (Windows Server 2008 R2 Foundation)
LAN interface 1 (192.a.b.y /24) connects to the internal network, 3cx, and the internet
LAN interface 2 (10.a.b.y /24) connects to our VoIP providers network.

Service provider
LAN Interface 1 (10.a.b.z /24)

All systems will use port 5060

As the 3CX machine will need to contact the PBXMate machine on the address 10.12.44.10:5060 the following route needs to be added on the 3CX machine, from the command line:

        route –p ADD 10.a.b.y MASK 255.255.255.255 192.a.b.y

The PBXMate machine acts a router between the 192.a.b and 10.a.b networks, so enable routing from regedit:

Computer\HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\services\Tcpip\Parameters
Change the value of IPEnableRouter from 0 to 1.
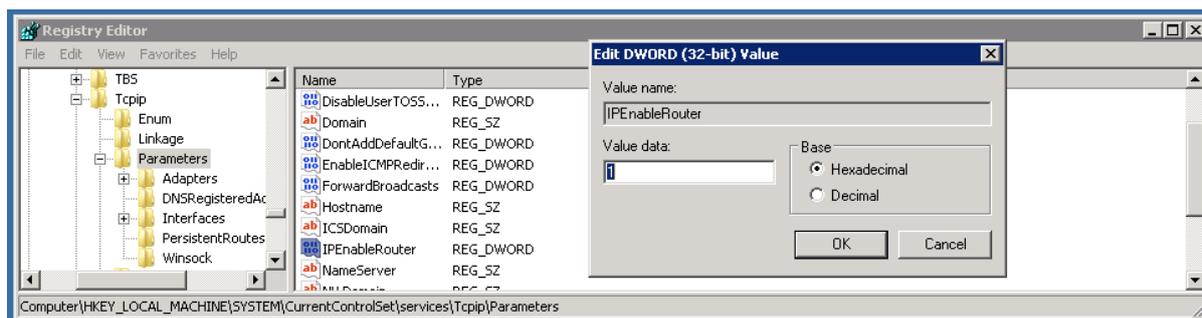
2: Setting up the trunk in 3cx

I have an existing trunk to my service provider, so I will just make the following modifications to it under the:

 "General" tab:
SIP Server Hostname - the IP address that PBXMate will accept requests on
SIP Server port – the port number that PBXMate will accept requests on
Outbound Proxy Port – the port number that PBXMate will accept requests on

"Advanced" tab:
Require registration for: - you do not need to register for the PBXMate trunk
Which IP to use in 'contact' field for registration – specify your internal LAN facing IP address

All Other settings remain unchanged, as DID Routing, inbound and outbound parameters still apply in the same way

That's all the changes needed to the trunk on the 3cx side. If you want to set up a new trunk, set it up as a generic SIP trunk.

3: setting up the trunks on PBXMate

Download and extract the PBXMate software to a location on the target server that will run PBXMate, for example C:\program files\solicall

Read the SoliCall PBXMate – Users Manual.pdf and set up the software, in general it goes something like so:

1 Add a SOLICALL_BIN system variable
2 Modify the pbxmate.conf file
3 Create the pbxmate_sip_routing.conf file

1 Add a SOLICALL_BIN system variable
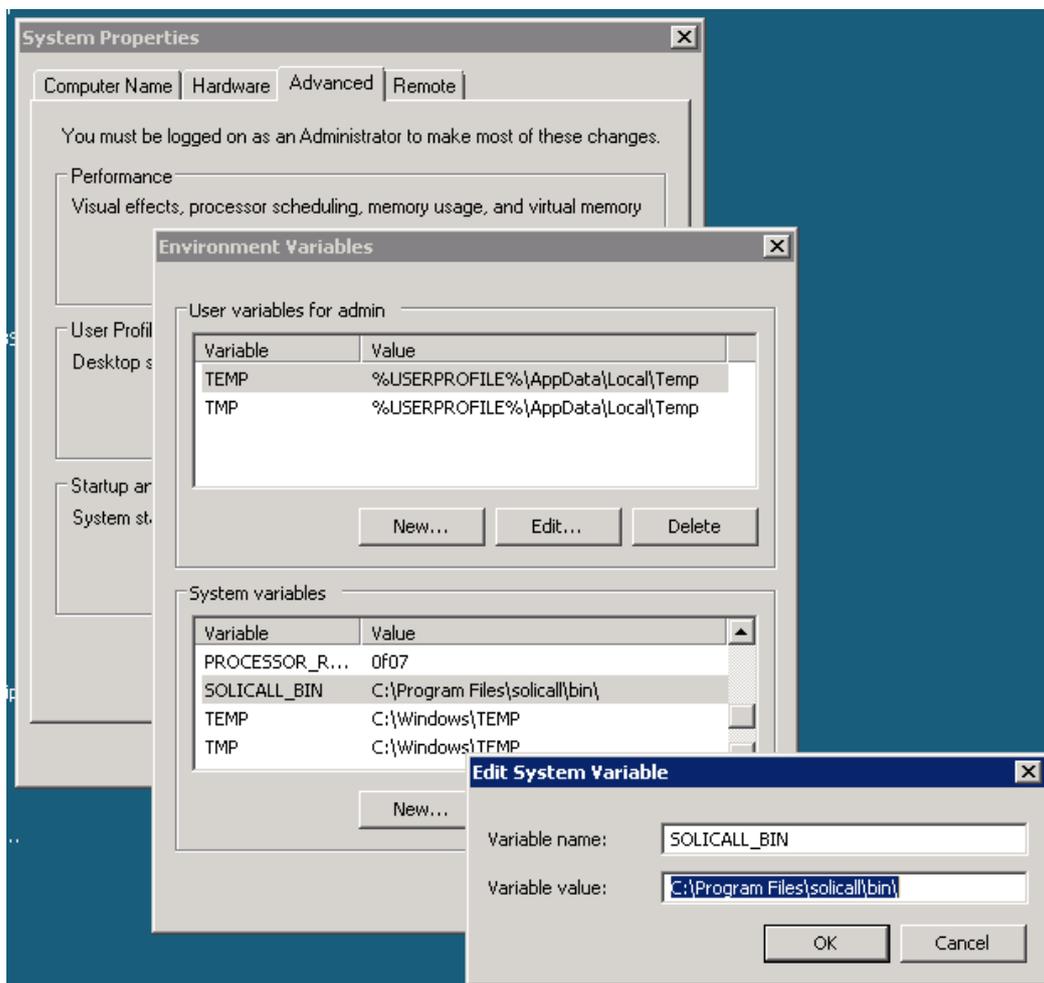
2 Modify the pbxmate.conf file, and create the pbxmate_sip_routing.conf

I have removed comments in the original pbxmate.conf, and replaced some of them with my own, this is to make this walkthrough easier to read. The actual pbxmate.conf file has more comments and there is more information available from the Solicall documentation.

Find the pbxmate.conf in the bin directory, open it in notepad, and edit the contents:

```
; ###################### Main Section #########################
ConfigVersion=2

IPVersion=4

; The details to connect to your VoIP Service Provider
SipProxyIP=10.a.b.z
SipProxyPort=5060
SipProxyDomain=10.a.b.z

; The Details of the PBXMate
SipUAIP=10.a.b.y
SipUAPort=5060

RtpUAPortInitVal=4000

Separator=--

; CmdBasedOnCallerID=1??@192.168.1.241=*--3;2??@192.168.1.241=*--1

DefaultCmd=*--3

TrunkName=

; My VoIP provider does not require registration
RegisterTrunk=0

; Trial version maximum of 4 calls, 4 pairs = 8 phones
NumPhonesToAllocate=8

InvitePhoneBasedOnName=0

; don't need phones to register
; OrigPhonesName=SoliCallOrig0;SoliCallOrig1;
OrigPhonesNamePrefix=SolicallOrig
; PairPhonesName=SoliCallPair0;SoliCallPair1;
PairPhonesNamePrefix=SoliCallPair
RegisterOrigPhones=0
RegisterPairPhones=0

; My VoIP provider does not require registration for the trunk, so leave as is
Secret=welcome

; RegisterExpiration=3600

; DOSExpressions=100@internal;2??@internal;
; AOSExpressions=100@internal;23?@internal;

StartWithFilterActivated=1

; RTPTimeoutInSeconds=30
; RTPTimeoutWhenOnHoldInSeconds=300

CanHaveDifferentIPForDifferentMedia=0

; I needed this to make my VoIP provider accept calls from PBXMate
PairPhoneShouldImpersonateTheOriginator=1

SipRoutingFile=./pbxmate_sip_routing.conf
```

```
; Removed ############# Noise Removal Section ####################
; Removed ############# Echo Cancellation (AEC) Section ####################
; Removed ############# Automatic Gain Control (AGC) Section ####################
; Removed ############# Statistics Section ####################
; Removed ############# DTMF Section ####################
; Removed ############# Misc Section ####################

; Record files to compare the impact of the PBXMate
OrigRecordingLibrary=../rec/
CleanRecordingLibrary=../rec/

; LicenseFile=./mylicense
```

3 Create the pbxmate_sip_routing.conf file

The pbxmate_sip_routing.conf tells the PBXMate to route calls destined to the 3 number blocks (64 9 900 0010 - 64 9 900 0029 and 64 4 900 0040 - 64 4 900 0049) to the 3CX server, otherwise calls get routed to the VoIP Provider.

Create a plain text file named "pbxmate_sip_routing.conf" under the bin directory, contents as follows:

```
192.a.b.x::5060:: 649900001?@10.a.b.y; 64900002?@10.a.b.y; 644900004?@10.a.b.y;
```

4: Making Calls

So now you should be ready to start the PBXmate software and make some calls!

Open a command prompt, navigate to the bin directory and run solicall_gui.exe
Open a browser window and go to http://127.0.0.1:8080/solicall_gui.html
Click start and check for the following lines in the text box:

> 02/20/11 06:04:15 INFO in MySip::mainLoop, All phones are active

This means the PBXMate is ready to do its thing

Make some test calls and check the directory you specified for the recording to go, in my case
C:\Program Files\solicall\rec

Each call has 6 audio files associated with it; the "cleaned" inbound and outbound audio plus the "original" inbound and outbound and "synch" files.

The audio files are in raw format, I used cool edit pro to open and listen to the audio files (8000 sample rate, mono, 16bit).

A quick look at the before and after graphs:
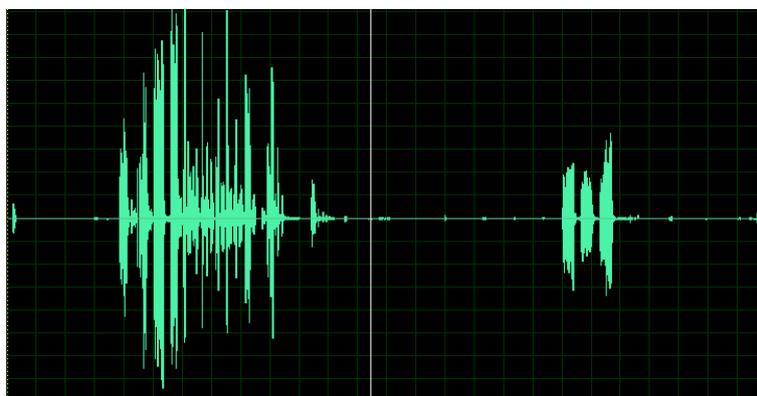
Graphs show outbound audio before and after they have been filtered.

4: Making Calls