Help me please

I have successfully created 2 container with OpenVZ on Centos 4.7 and successfuly installed the "Trixbox/Asterix" 2.0 application.

VZ1 has extension 1 to 5
VZ2 also has extension 1-5

VZ1 and VZ2 has internal private IP address and are sharing one public IP address

I can definitely access the internet from VZ1 and VZ2
but now I want to be able to register phones to VZ1 and VZ2 and I am unable to do that so far

I looked and read a few articles through the forum but no one actually stated step by steps on how they achieved their success

This is all new to me so please provided me with step by step instruction is possible

Any help is greatly appreciated

Thanks

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You have to create a trunk or a DUNDI link between the two Asterisk systems. Then you will need to create an outbound route to send the calls to the remote PBX systems via the Trunk. If you use DUNDI it will do it for you.

Jason

Thanks for the reply, but can you answer this question for me.

As mentioned I have a phone 555-1212 for Customer A pointing to public IP xx.xx.xx.x on VZ1
I Also have phone# 555-1313 for customer B pointing to the same public IP xx.xx.xx.x on VZ2

If both phone are pointing to the same public IP how will it know to go to their respective container since they are both using the same port 5060

Subject: Re: Register phones with same extension to multiple VE via internet
Posted by easyvoxbox on Wed, 01 Apr 2009 21:46:44 GMT
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If I am understanding you right, you have the same IP for multiple containers??

Each container must have its own IP. You can have private IPs for the containers with a NAT setup. In this case you will need to change the port number for the asterisk extension and then in the phone itself. You will also have to set up the port forwarding for the NAT using iptables. See this page for NAT setup:
http://wiki.openvz.org/Using_NAT_for_container_with_private_IPs

Subject: Re: Register phones with same extension to multiple VE via internet
Posted by ioscanner on Wed, 01 Apr 2009 22:05:06 GMT
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You need to setup an INBOUND Route. You will setup SIP to check for the DID that is in the request and forward that call to the correct extension, trunk or to another system as needed.

I currently support a few 100 people on one box. I am using a VE that syncs this to two different machines that I can move the VE to the second box if needed.

I am growing faster than a single asterisk box can handle so now I am doing SER that will handle all of the SIP reg and communicaitions and let Asterisk be the media servers. So I setup the same VE and let the SER pass the calls to the defined group of asterisk boxes for that inbound call or outbound call.

Jason

Subject: Re: Register phones with same extension to multiple VE via internet
Posted by msl7868 on Thu, 02 Apr 2009 20:38:42 GMT
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Jason,

Did you ever have a case where the DID does not get pass to the box. It could happen since that function really is up to the carrier to pass it.
In that case what do you think will happen?

thanks

Subject: Re: Register phones with same extension to multiple VE via internet
Posted by msl7868 on Thu, 02 Apr 2009 20:44:14 GMT
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No,

Each container has their own private IP but are sharing the public IP. Like you mentioned, I tried to have asterix run on diffrent port and I was having issues. Do you know of any good link with instruction on how to have asterix run on different port.

I am going to try both idea and let you know of the result

Thanks anyway for the feedback

Subject: Re: Register phones with same extension to multiple VE via internet
Posted by ioscanner on Thu, 02 Apr 2009 21:47:06 GMT
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It doesn't matter if you 200 different IPaddresses and 100 Asterisk boxes. You can connect them all together with trunks or DUNDI and create routes between the boxes. Each Asterisk box should know of the other boxes. DUNDI might be easier for you unless you use SER. SER give you a lot more control over the SIP traffic. If you plan to use IAX you will need to have them register direct to asterisk boxes. You can even have one config of asterisk for all of the boxes. But you would need to USE qualify= yes and write the logic in an AGI to find the extension on which ever PBX they are on. Or you could use Real-time asterisk and setup mysql so you can track down extensions to a box faster.

You might be better off using SER (SIP Express Router) then you can pass the calls to any of the asterisk boxes and use them just for the media stream. It would be much easier to manage and would scale better.

You can create a VE that has SER that listens to port 5060 and would route the calls to the Asterisk boxes to complete the calls based on LCR, Extension or what ever server handles that service.

There are many ways to do this. It might take longer to do the SER to many asterisk setup, but in the long run it would be the best solutions and allow you allot more flexibility. SER has a lot of
open-source users and is used by most VOIP providers. So it will scale the best. Asterisk sucks when trying to control the SIP and handle routing on a true SIP level. Also, you can product much better CDR records.

Jason