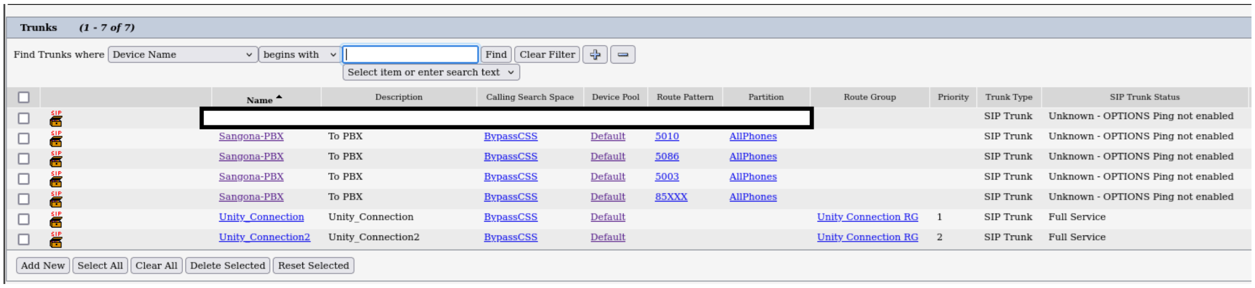
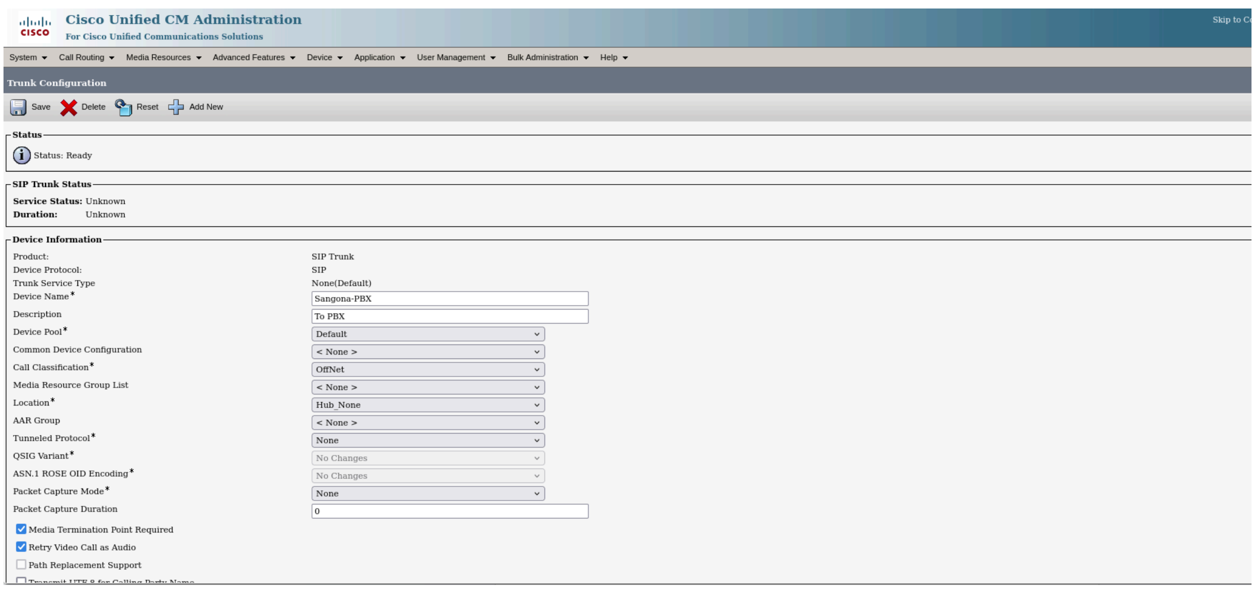
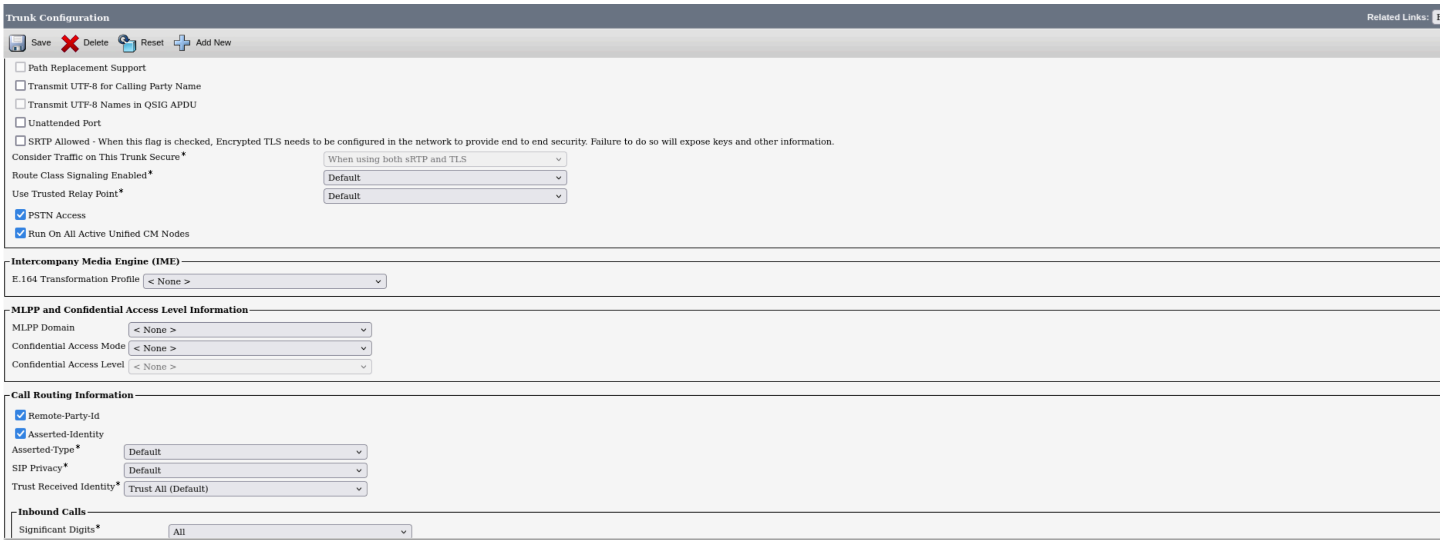


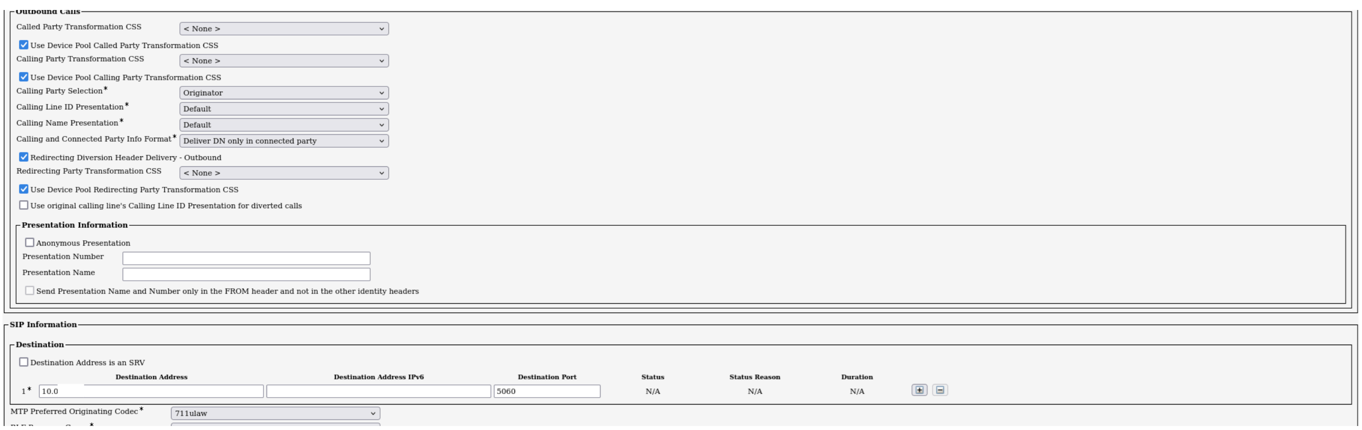
	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status
									SIP Trunk	Unknown - OPTIONS Ping not enabled
	Sangona-PBX	To PBX	BypassCSS	Default	5010	AllPhones			SIP Trunk	Unknown - OPTIONS Ping not enabled
	Sangona-PBX	To PBX	BypassCSS	Default	5086	AllPhones			SIP Trunk	Unknown - OPTIONS Ping not enabled
	Sangona-PBX	To PBX	BypassCSS	Default	5003	AllPhones			SIP Trunk	Unknown - OPTIONS Ping not enabled
	Sangona-PBX	To PBX	BypassCSS	Default	85XXX	AllPhones			SIP Trunk	Unknown - OPTIONS Ping not enabled
	Unity_Connection	Unity_Connection	BypassCSS	Default			Unity Connection RG	1	SIP Trunk	Full Service
	Unity_Connection2	Unity_Connection2	BypassCSS	Default			Unity Connection RG	2	SIP Trunk	Full Service

Unfortunately each extension you want to transfer is going to require a new sip trunk on the Cisco end. FreePBX you only need the one trunk. We have the standard 85XXX which will match all of our FreePBX extensions if you dial 8 then as we slowly move over extensions from one system to another we plan on doing dial pattern such as 500X, 501X. That will allow for 10 phones at once.

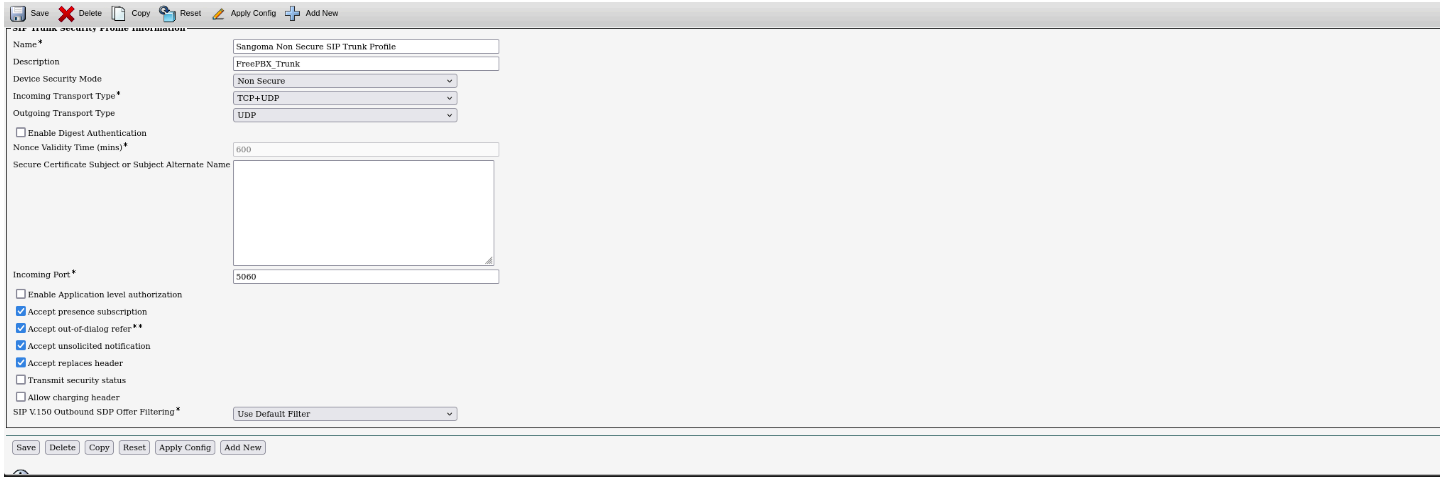


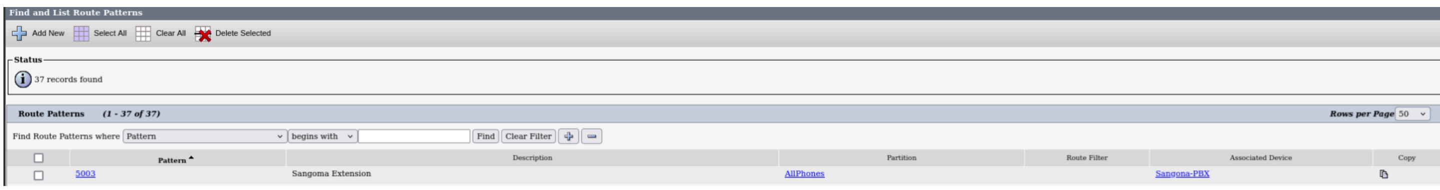
Make sure that your voice gateway is accessible from your Device Pool. This tripped us up for a while since we are not super savvy with Cisco Call Manager.





Codec is set to G711ULAW for the best possible compatibility with FreePBX. Other Codecs may work but G711 is the one that works best in our testing. Additionally make sure to set the DTMF signaling method otherwise it won't work.



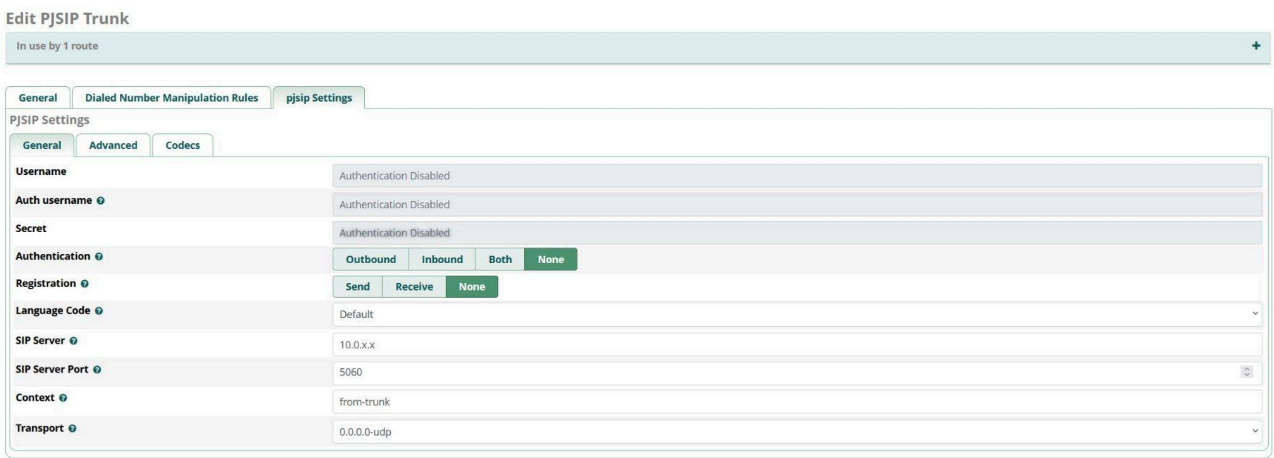


As I said before just have a route pattern so that the extensions you move can be natively called from a CUCM phone over to a FreePBX phone. If your fine with a steering digit you only need to do one route pattern. Make sure the CSS/Partition that the SIP trunk is part of can use your outbound dial patterns and just make sure to make equivalent ones in FreePBX and it should route out perfectly fine.

I believe that is all on the Cisco side of things. Dialed Number Analyzer is your best friend.

Please note that IVRs will break. We have made the decision to move the IVRs over to FreePBX early. If you use misc destinations in FreePBX you can set it to route the calls over the CUCM for any remaining extensions on CUCM. If you try and dial an extension while within the FreePBX IVR it will only go to a FreePBX phone there is no way that I know of to make it route to CUCM.

FreePBX side config for the trunk.



Make sure your Match Permit covers literally any IP that CUCM may use to communicate with your FreePBX. We had to make sure to put both of our UCM servers in the match permit as well as our voice gateway. As well as the WAN address just to be safe.

```
[from-internal]
include => Cisco-Caling



[Cisco-Caling]
exten => _5XXX,1,NoOp(Starting custom failover for extension ${EXTEN})
 same => n,Dial(PJSIP/${EXTEN},20)
 same => n,GotoIf($["${DIALSTATUS}" = "CHANUNAVAIL"]?outbound)
 same => n,Hangup()

 same => n(outbound),NoOp(Routing ${EXTEN} through Cisco trunk with 9 prefix)
same => n,Dial(Local/9${EXTEN}@from-internal,30)
 same => n,Hangup()

exten => _5XXX,n(outbound-failover),NoOp(Extension ${EXTEN} does not exist - failover
outbound)
 same => n,Dial(Local/9${EXTEN}@from-internal,30)
 same => n,Hangup()
```

Custom Config to put in Extensions_Custom

We decided to make it so that you have to dial 9 to get to the Cisco line. To prevent problems we made this script that can tell if there is no phone with that extension available and it will send it over to Call Manager.