FreePBX
TECHNICAL DOCUMENTATION

fusion

CONTENTS:

PURPOSE, SCOPE, AND AUDIENCE

The technical application note describes the configuration of asterisk based PBX including FreePBX. This Document is suitable for use by anyone deploying the Cloud Voice Service in conjunction with the GUI Based FreePBX. This document has a technical audience in mind – specifically IT Professionals skilled in Linux with experience in PBX administration and familiarity with VoIP technologies. This document is not for business administrators of people in other non-technical careers. In order to deploy the Cloud Voice Service, it is recommended you have the following skills:

- Understanding of UNIX or Linux operating systems, including:
  - Understanding of file and directory structure on target OS
  - Understanding of firewall configuration on target OS
  - Understanding of network configuration on target OS
  - Understanding of service configuration on target OS
- Familiarity with network troubleshooting tools, including:
  - Wireshark/Ethereal
  - dig/nslookup
  - ping
  - traceroute
- Familiarity with PBX systems, including:
  - Trunk configuration
  - Calling plan configuration
  - Extension configuration
  - Mailbox configuration
- Familiarity with Session Initiation Protocol (SIP)
- An understanding of all seven layers of the Open System Interconnection (OSI) model
- A complete understanding of your internal network structure, Network Address Translation (NAT) setup, and firewall setup
- A complete understanding of your public Internet connectivity

Fusion Connect can only provide support for FreePBX to the extent covered in this Technical Application Note. If your level of technical expertise does not include the above skills it is recommended that you obtain the service of a software professional.

DEPLOYMENT INFORMATION

Before you begin deployment, please locate the following information.

- External IP of Device: _____._____._____._____
- Maximum Calls Allowed: _____
- Preferred Codec: ulaw g729
- Is Device NATd: Yes   No
- Is Device Behind Firewall: Yes   No
- Is there a Local Firewall: Yes   No

At the top of your Welcome Letter you should see the following:

Account Name: **Name**
Account Number: **99XXXXXXXXX**
Order Number: **XXXXXX-X**

In your Welcome letter, Fusion Connect provides IP addresses to which you may send calls and from which you will should be prepared to receive calls. FreePBX support both DNS A records and IP addresses. Both should be listed in your Welcome Letter and either can be used. This IP will be dependent on if the trunk is configured as Static or Registration. For deployment of the FreePBX we recommended that the trunks be set as registration.

Traffic Type	IP Address	Domain Name	Protocol	Port Range
SIP Registration	216.86.41.69	sip2.thevoicemanager.com	UDP and TCP	5060/5061
SIP Static	216.86.41.68	peer2.thevoicemanager.com	UDP and TCP	5060
RTP – Registration	216.86.41.69	N/A	UDP and TCP	10000 to 65535
RTP - Static	216.86.41.68	N/A	UDP and TCP	10000 to 65535

PREFERRED AND ALTERNATE CODECS

Please set your preferred codecs to prioritize G711 or G729. When configuring codecs, please keep in mind that G.711 μLaw (ulaw) consumes approximately 87.2 Kbps of bandwidth per simultaneous call. G.729 Annex B (g729) will consume approximately 31.2 Kbps of bandwidth per simultaneous call. Also, keep in mind that G.711 offers superior call quality when compared to G.729, but only if you have enough bandwidth to support all of your simultaneous calls.

INSTALLATION AND NETWORK CONFIGURATION

For assistance on installation please visit https://www.freepbx.org/ for Free PBX.

Network configuration will vary based on the internal network and specific of the client's needs. We advise contacting the persons responsible for the maintaining the client's network for any specific configuration needs or issues.

CONNECTING TO FUSION

To start configuration, please choose Add Trunk (Figure 1). Recommendations for Sip Trunking is to use chan_sip as the trunk.   For testing purposes chan_sip was used for all compliance testing.
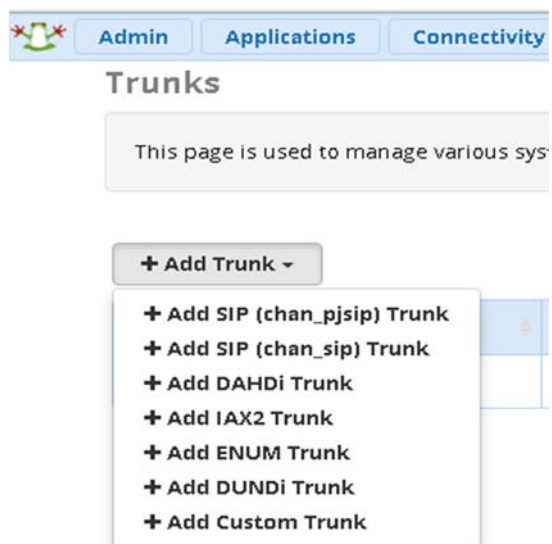

Figure 1

Under the General Tab please make the following changes:

1. Enter in the Trunk Name as with any preferred naming convention.
2. Set Hide Caller ID to No
3. Enter the Main Number in Outbound Caller ID in the following Format <XXXXXXXXXX>
4. CID options can be set to your preference
5. Enter assigned CCS in Maximum Channels
6. Asterisk trunk Dial Options can be set to your preference
7. Continue if Busy can be set to your preference. If no ARS routing is configured standard configuration is no
8. Set Disable trunk to No

Figure 2

Under the Dialplan Manipulation Rules please set your preferred outbound dialing rules. Please be advised that Fusion Connect requires 10 digits to be sent to our services to route calls. Please see an example of outbound dialplan used for testing purposes in Figure 3.

(prepend)	1	(NXXNXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1800NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1888NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1877NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1866NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1855NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(1844NXXXXXX	/	CallerID]	+🗑
(prepend)	prefix	(911	/	CallerID]	+🗑
(prepend)	prefix	(933	/	CallerID]	+🗑
(prepend)	1	(911	/	CallerID]	+🗑
(prepend)	9	(911	/	CallerID]	+🗑
(prepend)	91	(911	/	CallerID]	+🗑
(prepend)	91	(911	/	CallerID]	+🗑

Figure 3

Under the Sip Settings tab, Outgoing Tab, please configure the following:

1. Trunk Name. This can be any preferred name.
2. Under Peer details, please enter the corresponding information from your technical document as per Figure 4 for the host, username and password.

Figure 4

Under the Sip Settings tab, Incoming Tab, please configure the register string with in the following format as per Figure 5:
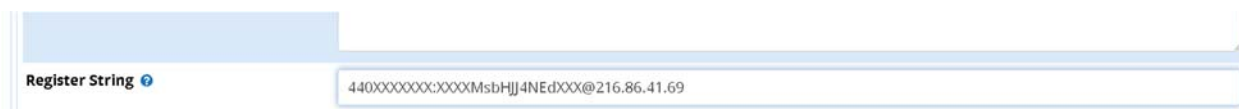
<username>:<password>@sipserver

Figure 5

Please be advised that to control individual CLID on outbound calls you must have the ability to send Remote Party ID, P-Preferred ID, or P-Asserted ID. For security purposes, the Cloud Voice Platform require registration trunks to send the BTN in the From Field on all outbound calls. Please consult the module package available for further information.