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Antenna base voice communication system at GMRT using Voice over Internet Protocol (VoIP)

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Background:

The existing antenna base voice communion system (phone), based on FSK modulation is more than 20 years old and also facing frequent maintenance. Recently as a part of the GMRT upgrade, the antenna base communication network was upgraded to TCP/IP (LAN). The FSK modulation communication system may be removed or will not be maintained in the near future. Hence it is the right time to upgrade the voice communication system.

Technology exploration:

Considering the antenna base voice communication requirement and the LAN availability at the antenna base, it was decided to opt for a “voice over internet protocol” (VOIP) system. The major issue here is that while upgrading any system at the GMRT, the system(instrument) has to go through the RFI testing. As per the observatory protocols, the new system should not generate the EM radiation in the observing bands. Before procuring the VOIP system, it was decided to conduct RFI test on these instruments i.e. VOIP handsets and select the particular model. For this RFI testing, four different handset models were brought to the lab for testing, namely Polycom, Grandstream, Coral and Panasonic. After the rigorous testing it was found that none of the above models were suitable for the GMRT. Shielding solution was there but it was very difficult to shield the VOIP handset. Due to RFI issues, we went for another option which was a VOIP adapter with analog handset. Adapter model was also tested for any possible EM radiation, this was too producing EM radiation. But then we finally decided to opt for an adapter, as it was easy to shield the adapter. Shielding box was made and the system tested which gave very good isolation.

Hardware:
Grandstream make VOIP adaptors (model: HT801) and Beetel make Handsets (model: B11) were procured for installation by Telemetry Lab. Shielding solution along with faraday cage, connectors and cables were provided by RFI team/Lab. Actual installation work at the antenna base was done by Telemetry Lab.

Grandstream HT801 specifications:

- Supports 1 SIP profile through a single FXS port and a single 10/100Mbps port
- TLS and SRTP security encryption technology to protect calls and accounts
- Automated provisioning options include TR-069 and XML config files
- Supports 3-way voice conferencing
- Failover SIP server automatically switches to secondary server if main server loses connection
- Supports T.38 Fax for creating Fax-over-IP
- Supports a wide range of caller ID formats
- Use with Grandstream's UCM series of IP PBXs for Zero Configuration provisioning
- Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi language prompts, flexible dial plan and more

Beetel B11 specifications:

- Ringer volume
- Tone pulse switchable
- Rdial
- Mute
- Flash
- Pause
- LED for ring indication
- Wall/desk mountable

Hardware installation:

As seen in the following images, hardware is installed into all 30 antennas, control-room, office reception and ngon-colony reception. At the antenna base, an adapter is installed inside the L2 network switch box, power and network connections are available inside the box. Finally the analog line (RJ11) for the handset is passed through the feedthrough to avoid any EM leakage, and the headset is installed on the wall for easy access.

Fig (1): Grandstream VOIP Adapter (HT801)

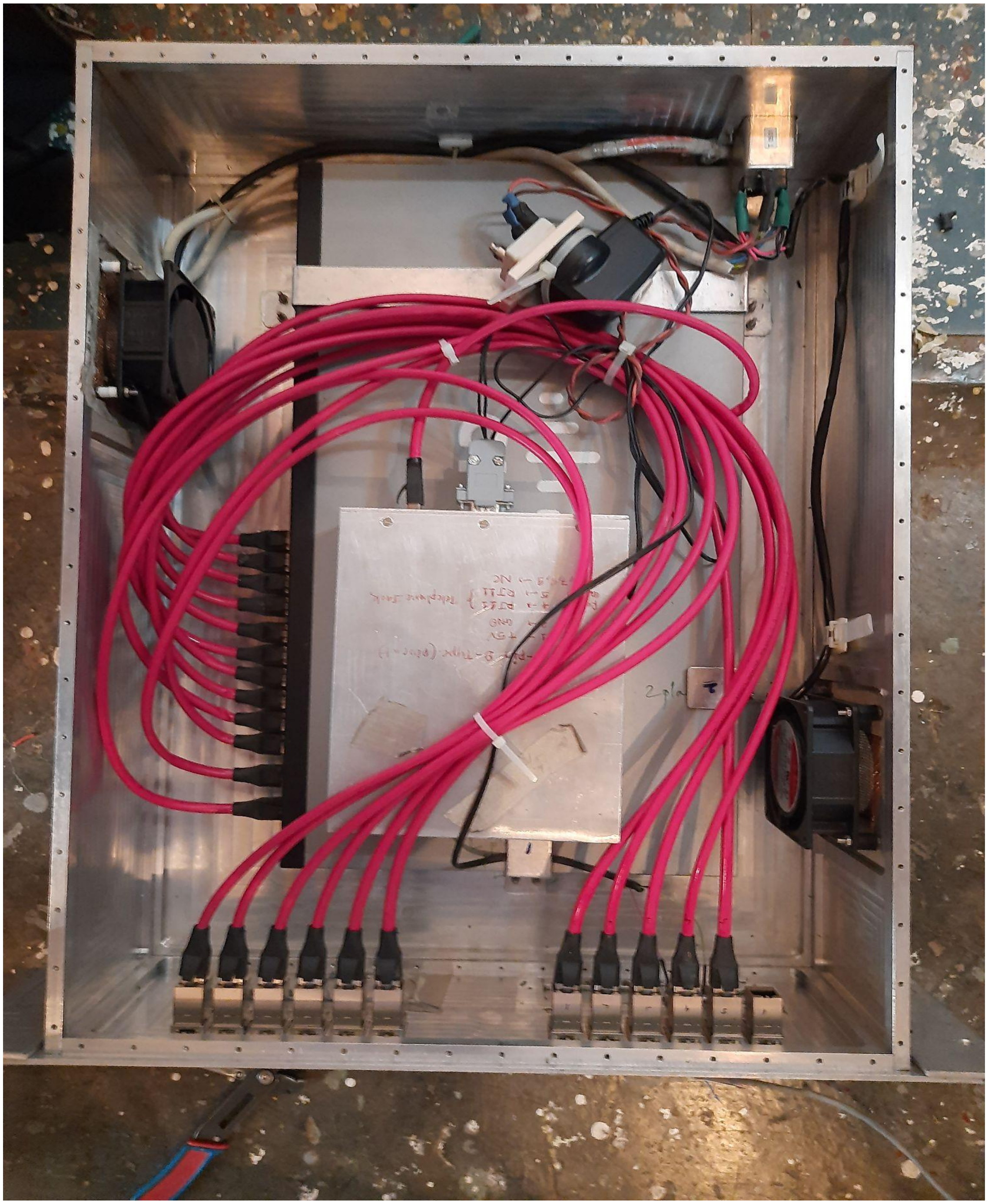

Fig (2): VOIP Adapter inside the double shielded box (switch box)

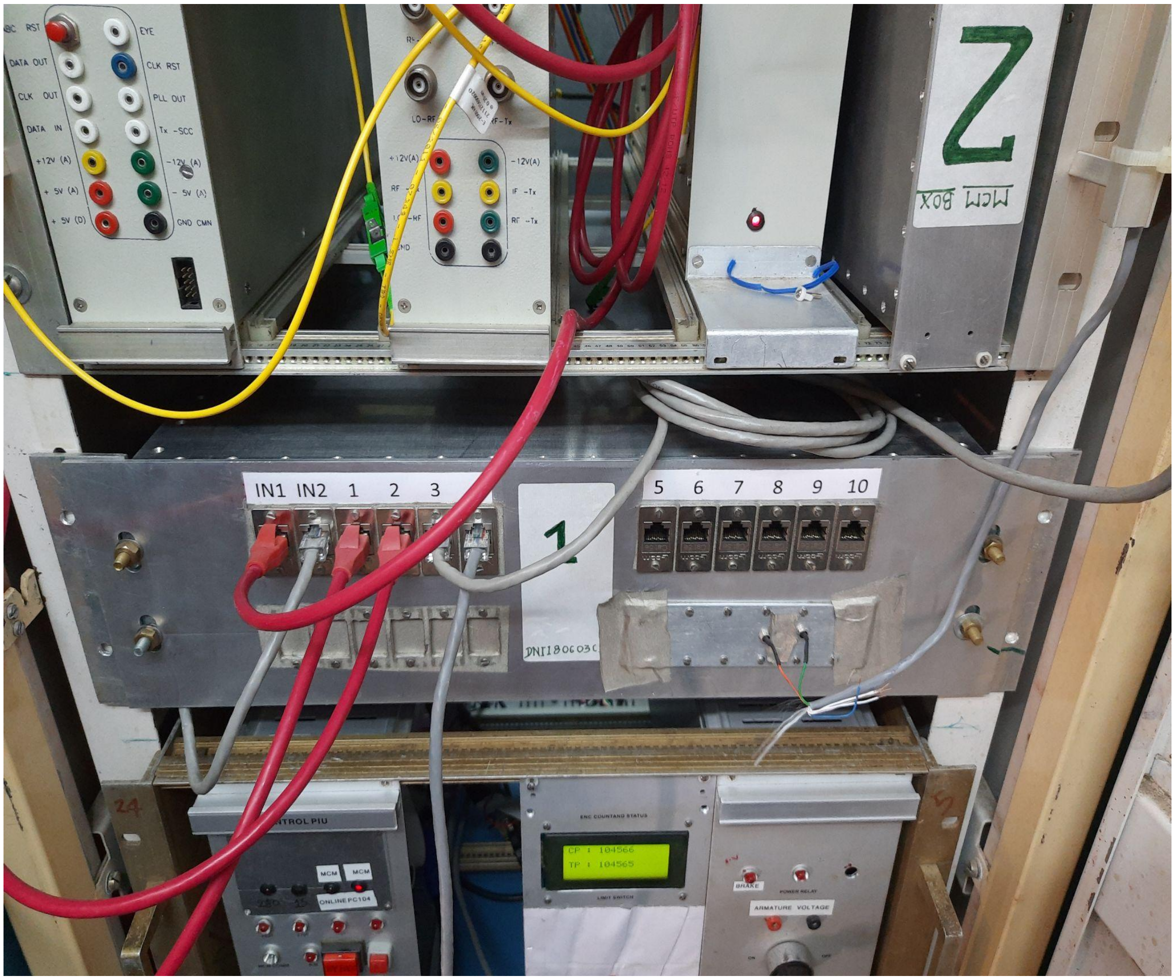

Fig (3): Shielded network switch (L2) box, adapter is inside the box

Fig (4): Beetel handset mounted on the wall

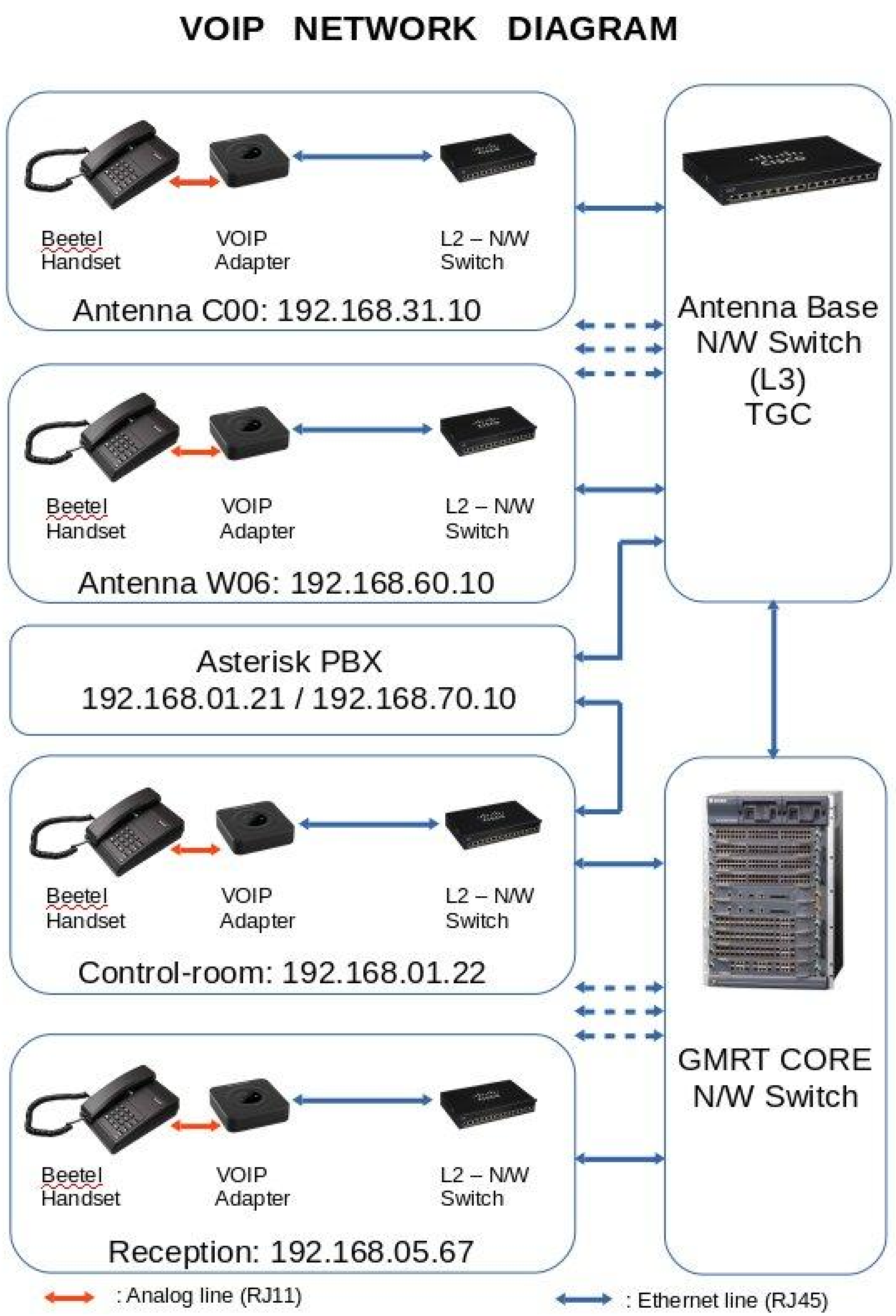

Fig (5): Network connection diagram

VOIP server (software):
We used the open source VOIP server to handle the voip communication. This server is installed in one of the TGC machines (ubuntu).

Asterisk:
Asterisk is an open source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. Asterisk powers IP PBX systems, VoIP gateways, conference servers and other custom solutions. It is used by small businesses, large businesses, call centers, carriers and government agencies, worldwide. Asterisk is free and open source. Asterisk is sponsored by Sangoma.

Today, there are more than one million Asterisk-based communications systems in use, in more than 170 countries. Asterisk is used by almost the entire Fortune 1000 list of customers. Most often deployed by system integrators and developers, Asterisk can become the basis for a complete business phone system, or used to enhance or extend an existing system, or to bridge a gap between systems.

Server configuration (asterisk):
It is installed on one of the TGC machines as follows
"sudo apt-get install asterisk" . After installation configured as follows.

Phone numbers:

C00 : 8500	C11 : 8511	S02 : 8532
C01 : 8501	C12 : 8512	S03 : 8533
C02 : 8502	C13 : 8513	S04 : 8534
C03 : 8503	C14 : 8514	S06 : 8536
C04 : 8504	E02 : 8522	W01 : 8541
C05 : 8505	E03 : 8523	W02 : 8542
C06 : 8506	E04 : 8524	W03 : 8543
C08 : 8508	E05 : 8525	W04 : 8544
C09 : 8509	E06 : 8526	W05 : 8545
C10 : 8510	S01 : 8531	W06 : 8546

Office Reception : 8600
N-Gon Colony Reception : 8700
Control Room : 8624
Telemetry Lab : 8630

/etc/asterisk/sip.conf

```
[general]
context=default
allowoverlap=no
udpbindaddr=0.0.0.0
tcpenable=no
tcpbindaddr=0.0.0.0
transport=udp
srvlookup=yes

;control-room
[8624]
type = friend
host = dynamic
hassip = yes
secret = 8624
```

```
;telemetry lab
[8630]
type = friend
host = dynamic
hassip = yes
secret = 8630


;gmrt-reception
[8600]
type = friend
host = dynamic
hassip = yes
secret = 8600


;narayangaon housing colony reception
[8700]
type = friend
host = dynamic
hassip = yes
secret = 8700


;C00
[8500]
type = friend
host = dynamic
hassip = yes
secret = 8500



;C01
[8501]
type = friend
host = dynamic
hassip = yes
secret = 8501
```

```
;C02
[8502]
type = friend
host = dynamic
hassip = yes
secret = 8502
.
.
.

;W06
[8546]
type = friend
host = dynamic
hassip = yes
secret = 8546
```

/etc/asterisk/extensions.conf

```
[general]
static=yes
writeprotect=no
priorityjumping=no
autofallthrough=yes
clearglobalvars=no

[default]

;control-room
exten => 8624,1,Dial(SIP/8624,45)

;tel-leb
exten => 8630,1,Dial(SIP/8630,45)

;gmrt-reception
exten => 8600,1,Dial(SIP/8600,45)

;narayangaon housing colony reception
exten => 8700,1,Dial(SIP/8700,45)
```

```
;C00 -
exten => 8500,1,Dial(SIP/8500,45)
exten => 8501,1,Dial(SIP/8501,45)
exten => 8502,1,Dial(SIP/8502,45)
.
.
.
.
exten => 8546,1,Dial(SIP/8546,45)
```

Client configuration (VOIP adaptor):

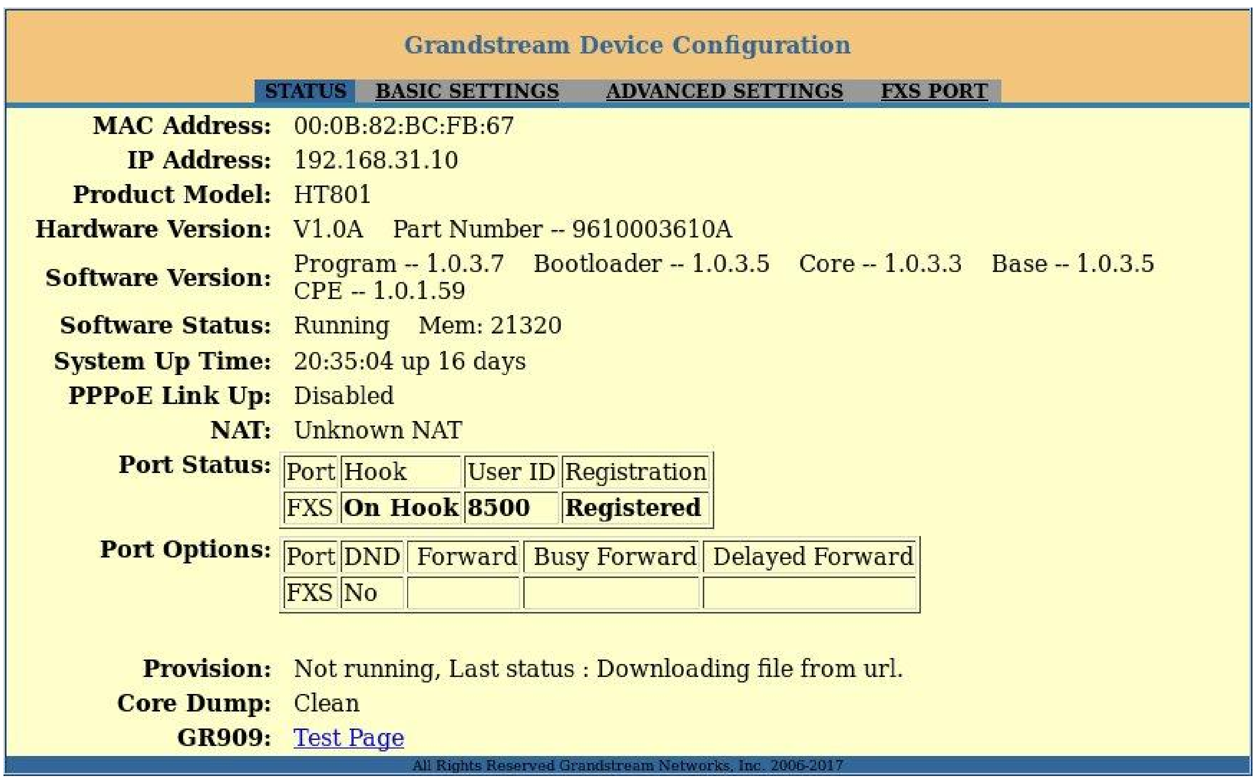

Fig (6): Web interface for Adapter

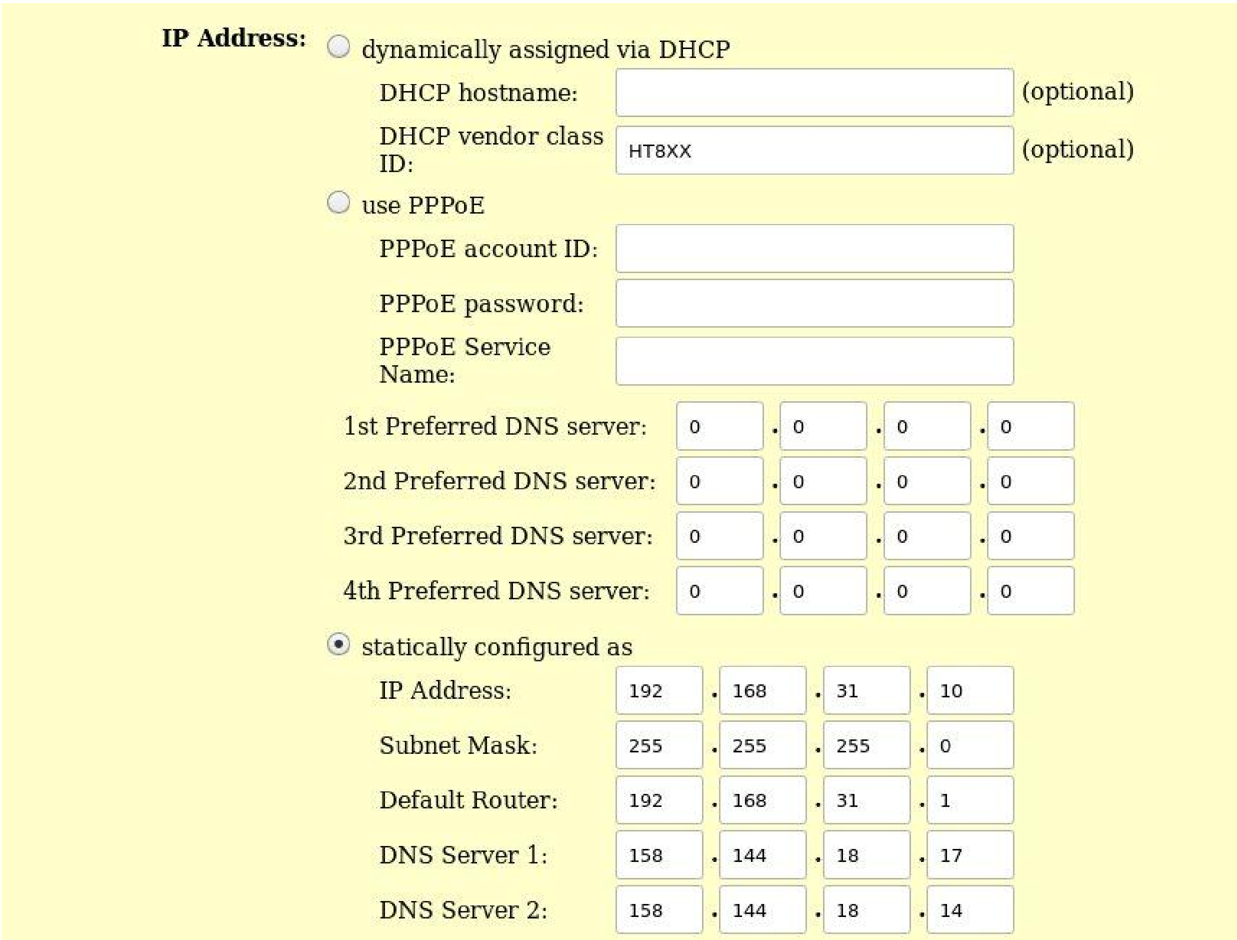

Fig (7): Static IP address config for Adapter

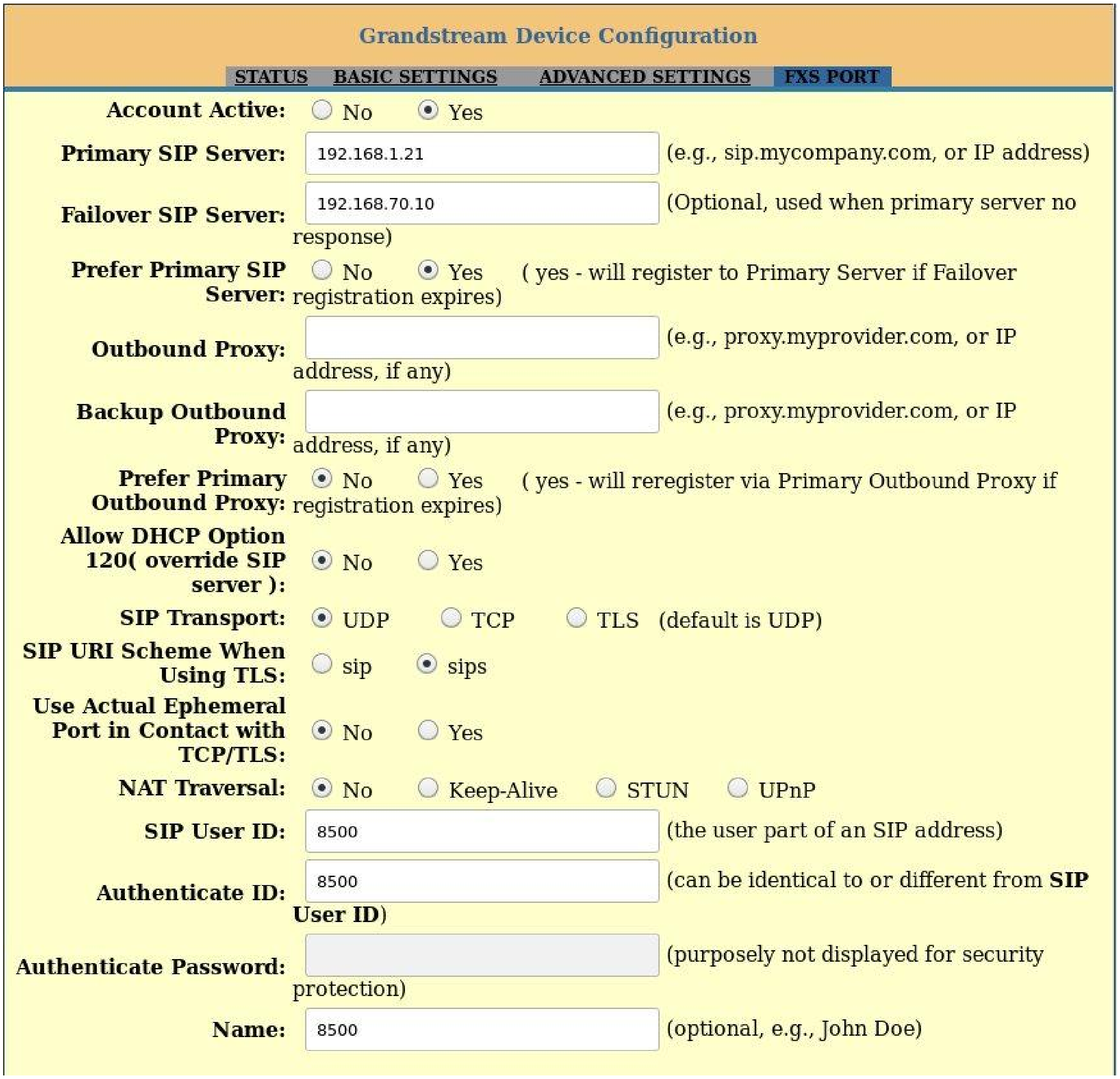

Fig (8): FXS port settings for Adapter

Training:
Required training is given to the staff members for hardware installation and software configuration.

Future Plans:
There are two telephony systems in the office, the first one is for the office and it is maintained by an electrical group. The second one(new) is for the antenna base and it is maintained by a telemetry group. Earlier it was not possible to combine or merge these two systems but now due to the upgrade of the antenna base telephony system it is possible to combine or merge these two systems. After this change one can call from any office lab to the antenna base and vice versa. In the near future we have plans to merge these two systems.

Conclusion:
The combination of VOIP adapter (Grandstream) inside the shielded box and analog handset (Beetel) gives good EM isolation. Installation and configuration of all the units was completed by the mid of February 2021. It has been four months since installation and the system is working very smoothly with excellent voice quality. So far we have not faced any problem or maintenance with the system.

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Software Configuration : Raj Uprade, Bhavesh Kunabi, Amol Chouhan.
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RFI testing and shielding solution: Pravin Raybole and team.
Network Configuration : Mangesh Umbarje and team.
Users: Control-room team.

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