VoIP Exchange - A Scheme for IR

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Abstract

Indian Railways has a large Railway Telephone Network (RTN) covering all its zones, divisions, production units, CTIs etc. It also has its own STD codes for intra exchange routing of calls. In the year 2009, the core of the RTN was shifted to IP by establishing an NGN[1] using the MPLS[2] infrastructure of RailTel Corporation of India Limited (RCIL).

In the year 2012, Railway Board issued guidelines for using VoIP exchanges carving a path for full migration of RTN on VoIP. The instructions have since been modified in the year 2015 to facilitate Railways in this migration. A few of the Railways have installed VoIP exchanges and are already reaping the benefits of the new technology. It is however seen that most of the Railways are struggling to adopt this new technology.

This article is an attempt to present a scheme that can be used in Railways for adopting VoIP exchanges that are distributed, simple and can scale horizontally.

1 Introduction

Indian Railways uses its own Telecommunication network to fulfill its voice communication requirement between zonal headquarters and its divisions. For this purpose, Railways has created Next Generation Network (NGN) using RailTel's MPLS infrastructure. The general connectivity of the Railways Telecom Network is shown in figure 1.

The zonal and divisional exchanges are connected to the NGN using E1/PRI. The NGN consists of soft-switch in 1+1 redundancy mode. The main switch is at Delhi and the secondary switch is at Secunderabad. The soft-switch is at the heart of NGN and controls dialplan as well as call routing. NGN also consists of media gateways distributed across India. The media gateways are used for connecting various exchanges of Indian Railways to the NGN. Media gateways and soft-switch are connected to each other on IP using the MPLS network as a Layer-3 MPS VPN.

[1]Next generation Network
[2]Multi-Protocol Label Switching

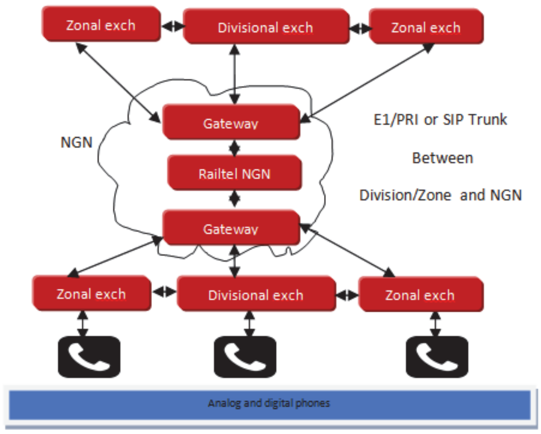

Figure 1: Railways NGN Architecture

2 Policy Guidelines

In the vision 2020 document of Indian Railways, Voice network modernisation has been envisaged. Accordingly, the core of the Railway Telephone Network was upgraded to NGN. In Dec 2015, Board has issued a policy guideline for adoption and migration to VoIP in the zones and divisions of Indian Railways (Telecom Circular No. 12/2015). Salient features of this policy guidelines are as under:

a) For adopting IP telephony in Indian Railways, SIP protocol should be used.

b) Existing Railnet will be used for IP Telephony as a backbone.

c) IP Exchanges based on **Open Standards** should be used.

d) Common IP telephony infrastructure should be enabled for Intercom and main railway number.

e) FXS gateways should be used for connecting FAX machines and FXO/PRI gateways should be used for connecting PSTN.

f) PoE switches should be used in Railnet and PoE should be used to power the IP phones.

3 Railways PBX Requirement

Railway PBX are required to provide the facility of Intercom, Railway phone as well as PSTN connectivity, all integrated into one phone. Let us see these requirements in some details.

3.1 Intercom

Each of the departments in a division/zonal office is provided with an Intercom phone. This phone is with two or three digit number. It is used for easy communication within a department. It is quite possible to have the same Intercom number in different departments. Calling an Intercom number from one department to the Intercom number of another department is not allowed. The Intercom numbers are direct numbers and boss-secy arrangement is not required for Intercom numbers.

3.2 Railway Phone

Each phone of the IP exchange shall be provided with a railway number. A railway number is a five digit number that is used across Indian railways in the Railway Telecom network. Each of the Railway exchange is also provided with a three digit STD code. Thus a railway number will be an eight digit number. These numbers may be provisioned with boss-secy facility or it may be direct.

3.3 PSTN Connectivity

It should be possible for each of the phones to be provided with a PSTN number. Direct-in-dialling should be used to define virtual numbers to be routed through these PRI lines by the service providers. Again, these numbers may be provisioned with boss-secy facility or it may be direct.

3.4 NGN connectivity

It should be possible for the IP exchange to interface with the NGN using either PRI or SIP for connectivity to the Railways Telephone network for connectivity to other railway exchanges.

3.5 Caller Line Identification

As detailed above, it is seen that one phone will have three numbers defined - one for Intercom, one for Railway number and one as the PSTN number. Now, when a call is placed using a phone, the IP exchange should properly transmit the CLI to the called party. Hence, if an intercom call is placed, the intercom number shall be sent as the CLI. If a Railway number is called, the Railway number shall be sent as the CLI. Similar shall be the case for a PSTN number.

3.6 Boss-Secy arrangement

The Boss-secy arrangement is normally provided to day using the 1+1 plan phone. The basic requirement of this arrangement is as under.

a) When the number of boss is called, secy's phone shall ring. It should be possible for her to take the call and do a blind/attender call transfer to the boss.

b) In case the call is not answered by secy, the call may be sent to the boss or dropped as per configuration to be defined by the user.

c) The secy should have the information if the boss-phone is free.

d) When the secy is on phone, boss should have the provision to make a call.

e) It should be possible for the boss to give a ring/buzz to secy even when she is off-hook.

4 VoIP Exchange Architecture

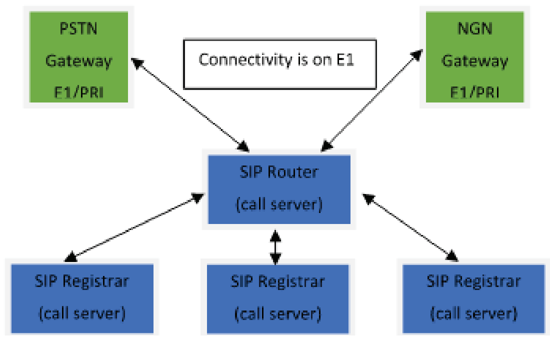

Figure 2: VoIP Exchange Architecture

Figure 2 shows the architecture of the VoIP exchange at divisional/zonal office. It shows an IP Exchange with SIP Router (Call Server) and SIP Registrars. The SIP Router in this case is a call server which takes the calls from SIP registrars, NGN or PSTN and route them to their appropriate destination. It maintains a centralised call-routing plan. It should be provided with redundancy as it is the heart of the IP exchange.

SIP registrars are the servers where the credentials of the IP telephones are defined. Its job is to authorise a SIP phone to register itself to access the services of the IP exchange. On a normal COTS[3] server we can have around 1000 phones registered. In case more than 1000 phones are required, then it is suggested to have another server for 1000 phones. Thus the architecture will scale horizontally.

The SIP registrars should work in redundancy. Two SIP registrars can be configured to work in static load-sharing fail-over mode.

The connectivity between SIP Router and SIP registrars is made by using the existing Railnet infrastructure, so no need to create a separate network.

Digital Media Gateways should be used to connect the IP exchange with the NGN and PSTN.

4.1 PSTN Connectivity

Figure 2 also shows the connectivity of the IP exchange with the PSTN. This is done using a media gateway on E1/PRI. It helps IP exchange subscribers to call PSTN as well as GSM network. The Dial plan to call out to PSTNs and GSM network is configured in Call Server itself.

4.2 NGN Connectivity

In figure 2, NGN connectivity is done on E1/PRI, which connects the IP exchange to Indian Railway Telephone Network. In future, this connectivity will be done on SIP Trunk. This will make an all IP voice network of Railways.

4.3 Redundancy

For redundancy requirement, the phones used in the system should have the provision to register on two SIP servers. One primary and the other secondary. The phone shall try to register on the primary SIP registrar failing which the secondary registrar shall be tried.

The SIP registrars shall also use a simple scheme for fail-over Let us take an example where we have 1000 phones. So, we create 1000 accounts on first registrar and the same 1000 accounts ion the second registrar. Now, in the first 500 IP phones, first registrar should be configured as the primary SIP server and second registrar should be configured a the secondary SIP server. Similarly for the next 500 phones, first registrar should be configured as the secondary SIP server and second registrar should be configured a the primary SIP server. So, if one SIP registrar goes down, the IP phones will get themselves registered to the other SIP registrar and will be able to provide the telephony services to the users.

The SIP router can also be duplicated with the SIP registrars trying to use the first SIP router for call forwarding failing which the other SIP router can be used. This can be programmed while call routing.

The gateways can also be duplicated accordingly and SIP routers can use on the gateway to forward calls. Here, however we have to see that the lines are also to be distributed over multiple gateways is a way to achieve redundancy.

[3]Common Of The Shelf

5 Benefits of IP Exchnage

When we start migrating from existing TDM exchanges to IP Exchanges, initially it will be required to run both schemes by interlinking. But after fully establishing the IP Telephony systems, it will function purely on IP network with IP devices and IP protocols.

a) We should be using SIP as the VoIP protocol; it is open source protocol more suitable for voice Transmission on Internet in real time.

b) Network unification IP telephony blends voice, video, and data by specifying a common transport, IP, for each, effectively collapsing three networks into one.

c) Simple IP phone system, just suitable to carry out day to day working of Railways.

d) It uses open standards and open protocols so there is no vendor-lock for any of its protocols.

e) Easy to maintain.

f) Common copper cable infrastructure for both voice and data (Railnet).

g) The 1+1 plan system, which is getting obsolete, will no longer be required as the boss-secy feature will be handled by the IP exchange in the dial-plan itself.

6 Conclusions

The telecommunications in Indian Railways is being revolutionized with the emergence of next generation networking (NGN). Next-generation networks with IP based Telephone Exchanges allow convergence of different network architectures. The frontrunner of this convergence includes SIP (Session Initiation Protocol) making it possible for triple play services (voice, data and video) to flow over the same network. Soft-switch is the key component of Telephone Exchange network and is designed to replace traditional hardware telephone switches by serving as gateways between telephone networks. Many concepts deployed in Railway telephone Network as explained in this paper have been translated into soft-switched networks, including signalling, voice codecs and transport. Soft-switch is disruptive to legacy networks. VoIP components are cheaper, simpler, smaller and more convenient to use than their predecessors in the network. In this article we have explained the IP PBX network architecture, protocols used in VoIP network, services that supports etc. The main purpose of this paper is to explain how to deploy VoIP Exchanges in Indian Railways. Here we have tried explaining specific requirements of Indian Railways and how to deploy them in proposed VoIP exchanges.

The information / views expressed in this paper is of the authors and are based on their experience. Comments / observations may be sent to the author at smhafeezali@gmail.com.

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Sri Rajesh Jhari has 27 years of experience in Indian Railways. He has worked in different makes of exchanges and has a good command on Coral Flexicom Series of exchanges. He is currently working as an Instructor in the Telephony Lab of IRISET. He has recently setup the VoIP exchange experiment setup with ten benches in the Telephony lab.