

VoIP Based Tele-medicine Call Center–Issues, Challenges and Proposed Solution

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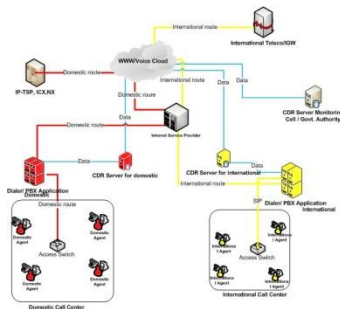
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Graphical abstract



Abstract

In recent years, medical call centers have started using IP telephony services to minimize the overhead telecom expenses. However, the advent of Voice-Over-IP (VoIP) technology has also created a major discontinuity in telecommunication sector due to illegal VoIP or gray calls, causing a great impact on the voice market. This brought few challenges to the countries' regulatory bodies. In this paper, we spell out one of the key challenges: in order to mitigate illegal VoIP calls, whether the regulatory body should allow IP telephony to be practiced for both domestic and international voice driven tele-medical consultation center operation combined or separately. We propose architecture and schemes for a medical call center. We also propose some guidelines and/or policies for both call center operator and the telecom regulatory authority. The proposed architecture and schemes are implemented in a pilot project basis in two phases and the test bed result is presented in this article.

Keywords: Illegal VoIP; medical call center; ILDTS policy; Tele-healthcare; SIP trunk; BTRC

Abstrak

Sejak kebelakangan ini, kebanyakan pusat panggilan perubatan telah mula menggunakan perkhidmatan telefoni IP bagi mengurangkan perbelanjaan pasti dalam telekomunikasi. Walau bagaimanapun, kemunculan teknologi IP atas suara (VoIP) dan juga VoIP haram atau panggilan kelabu telah memberikan kesan yang besar kepada pasaran suara. Ini telah menghasilkan beberapa cabaran kepada badan kawal selia sesebuah negara. Dalam kertas kerja ini, cabaran utama isu ini dibentangkan: bagi menyekat panggilan VoIP haram sama ada badan pengawal selia perlu membenarkan telefoni IP diamalkan oleh operasi berasaskan suara di kedua-dua pusat perundingan tele-perubatan tempatan atau antarabangsa secara bergabung atau berasingan. Kami cadangkan seni bina dan skema bagi pusat panggilan perubatan. Kami juga mencadangkan beberapa garis panduan dan/atau polisi bagi kedua-dua pusat panggilan pengendali dan penguatkuasa kawal selia telekomunikasi. Cadangan seni bina dan skema dilaksanakan dalam bentuk dua fasa projek rintis dan hasil ujian dibincangkan dalam artikel ini.

Kata kunci: VoIP haram; pusat panggilan perubatan; ILDTS dasar; tele-penjagaan kesihatan; batang SIP; BTRC

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1.0 INTRODUCTION

A. Call Center Background

A call center can be defined as a centralized office that can handle a large number of inbound and outbound telephone requests. In recent decades, there has been explosive growth in the number of companies that provide consumer services via the voice. These call/contact center technology based Business Process Outsourcing (BPO) mainly provides telemarketing and customer care processes. Developing countries like India and Philippine are pioneers in BPO sector. There are mainly two types of BPO exist: international and domestic business services. Most organizations with customer contact – private companies, as well as government and emergency

services – have reengineered their infrastructure to include from one to many call centers, either managed internally or outsourced [1-2], [9]. Voice over Internet Protocol (VoIP) enables the Internet to be used as the transmission medium for phone calls where voices are sent in packets using IP. In this paper, we study VoIP based call center issues in developing countries. Our focus is more on the developing world where we consider Bangladesh as a case study region.

Tele-medical consultation through telephony system concept is new trend in tele-health era in Bangladesh. Grameen Phone is the largest mobile operator in Bangladesh, first established a Global system for Mobile (GSM) based medical call center named HealthLine 789 in 2006 [12]. The deployment cost of this first medical telephony project was very high because of circuit

switching based devices [12]. Subsequently, different medical center, hospital and telecom operators have started GSM and circuit switch based telemedicine. In that time there was absent of IP telephony technologies and call center guideline [13]

Bangladesh Telecommunications Regulatory Commission (BTRC) has started allowing setting up of call centers in commercially in April 2008. However, such business started springing up this year. After receiving their licenses only three and a half years ago, more than 80 per cent of the country's call centers have gone out of business. This has created a void in the potential sector, which was supposed to earn a considerable amount of foreign currency every year, leaving it unutilized and unexplored [8-10]. According to BTRC, licenses were issued to 426 companies in 2008 under the categories of call centers, hosted call centers and call center service providers. Out of them 337 have surrendered their licenses and do not operate any more. Among the rest, only 47 international and 19 local call centers are currently in operation. Last year, 55 international and 17 local agencies were operating. Incidentally, steps have been taken by the authorities to fulfill three out of four demands that the BTRC in a report had placed before the parliamentary committee of Ministry of Post & Telecommunication (MoPT) in June 2010 to enhance the business. Government, MoPT and BTRC have jointly taken some steps to reduce the cost of Bandwidth, continuous power supply to the call centers and developing human skill in spoken English. However, business of international voice BPO is not so successful in Bangladesh. International call center provider faces many issues, such as absence of redundant transmission cable (e.g., submarine cable), unavailability of skilled agents, and scam business [8-14]. As a result, the recent call center business in Bangladesh is not moving forward. Especially, international call center operations are not able to make money though the domestic BPO are doing well. Moreover, sometimes international voice BPO entrepreneurs also face some legal issues. For example, as per Call center guideline domestic and international call center are not supposed to establish in same premise [2], [8], [29].

Recently (2009-2012), most of the local telecom operators and hospitals outsourced their medical tele-consultation and promotion process to different commercial call centers. Even International medical processes are also outsourced in Bangladeshi call center now a day. In this article, we consider the international route (both inbound and outbound) is international tele-medical call flow. Subsequently the tele-medical call flow from local is considered as domestic route.

Recently call center industries in India, Pakistan, Bangladesh, China and Philippines has multibillion dollar business. As per telecom regulatory of these countries except Bangladesh, there is no issue with international and domestic call center at same premise though illegal VoIP is prohibited without proper Voice Service Provider (VSP) licensing. And VoIP is not likely illegal there. Because their telecom and data transmission authorities or regulatory having a common platform for different telecom services. Moreover, the total call center industry under the same umbrella network and which is easy for their local telecom authority to monitor the voice route. As a result, they have many local licensed IGW for VoIP operation [28-30].

Recently South Africa trying to explore the call center business but due to high wages and accent barrier they are not doing well in call center business. And their telecom regulatory having no VoIP port issue like Bangladesh [2], [4], [16]. BTRC provides a traditional design (Please see Figure 1) for call center model, however that is only for international call operators, no clear indication for the domestic call operators [3], [8], [28-29].

On May 4, 2011, a meeting held with BTRC and Bangladesh Association of Call Centre and Outsourcing (BACCO) and the only one agenda was regarding that issue. In the meeting, the first author

of this paper (on behalf BACCO) was presented a technical solution that overcomes the main problem "Illegal VoIP termination issue".

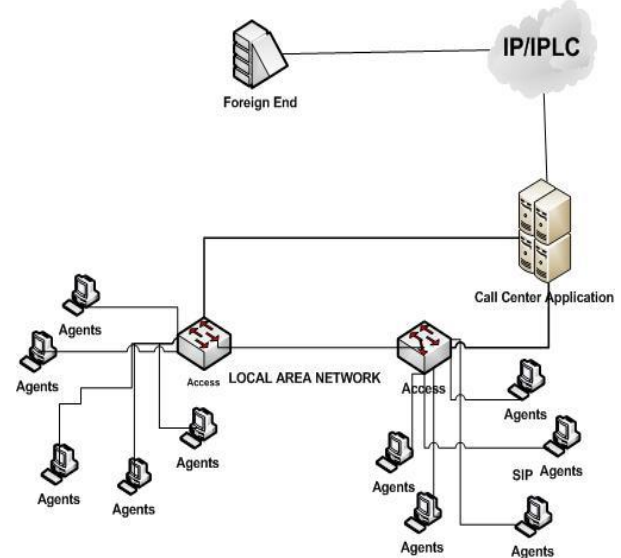


Figure 1 Call center architecture as per regulatory guideline^{3,8}

B. Illegal VoIP Issues

Voice over Internet Protocol (VoIP) is a much-talked issue in the Information Communication Technology (ICT) sector, especially in Bangladesh. It is generally believed that overseas call termination using VoIP started during 2000 when the use of VSAT (Very Small Aperture Terminal) was liberalized with the aim of promoting software export. Until Bangladesh Telecommunications Regulatory Commission (BTRC) formed in 2002, the regulatory functions of telecommunications sector had been overseen by the Ministry of Post and Telecommunications (MOPT). Under the guidance of MOPT, the then government operator Bangladesh Telecom Company Limited (BTCL) led actions against illegal voice termination using VoIP [1-3].

After BTRC came into operation, it has been hunting for the effective solution to curb overseas call termination using VoIP technology through employing foreign consultants, conducting workshops and dialogues with the stakeholders. However, no consensus was made about legalizing overseas voice communication using VoIP which could be acceptable to all quarters. Through an initiative to open up VoIP in 2007, International Long Distance Telecommunications Service (ILDTS) Policy 2007 came in the telecom sector with a layered concept. A three-layer structure for voice communications namely International Gateway (IGW), Interconnection Exchange (ICX) and Access Network Service (ANS) operator, National Exchange (NX) and two-layer for data communications namely International Internet Gateway (IIG) and Internet Service Provider (ISP). For international call center BTRC provides one port open IP for voice transmission outside Bangladesh. It is observed that blocking IP addresses is also not the effective technical way because as soon as one or some IP addresses are blocked illegal voice operators can switch over to different IP addresses. They might have handsome numbers of IP addresses and different PBX port in their hand [3-7].

C. Call Center Technology Use for Illegal Voice Termination

Mainly call centers run on IP based telephony system. There are another type of call center exists which is operated by legacy PBX, TDM or ISDN network [11, 12]. For international BPO operation, BTRC allows either IPLC (International Private Lease circuit) or IP based voice transmission. However, the IPLC technology is very expensive and it is not viable for call center operation. Thus the call center operators choose IP technology. Moreover, BTRC allows one SIP port open IP for each call center for international operation by which it is possible to make illegal VoIP calls. Illegal VoIP means that only the call termination from the foreign IGW. In Bangladesh, only the telecom operators have reserved the facility for call origination and it is not cost effective to setup call shop or calling card system [2-5], [8-9].

By using call center IP it could be very cost effective for doing VoIP since termination of call treated as international inbound BPO service where the foreign IGW has to pay incoming call. Any call center operator is able to terminate the commercial incoming call through IP telephony service providers or using the GSM SIM bank. In addition, the cost involved is only the local telecom charges for the IP foreign call termination to local mobile or landline phone.

The technological methodology could be either by Internet Protocol Telephony Service Provider (IPTSP) or by using the GSM gateway. In the case of IPTSP, voice routes from end directly go to call center application IP-PBX and from as per the dial plan the voice re-route to IPTSP end to terminate the call to local telecom network exchange/GMSC through interexchange carrier (ICX). On the other hand, GSM SIM bank based voice termination issue, calls are re-route to SIM/channel bank for terminated in GSM/CDMA/PSTN network. The both cases are called grey voice termination and not legal voice operation as per telecom regulation authority (e.g. FCC, Ofcom, BTRC, DoT).

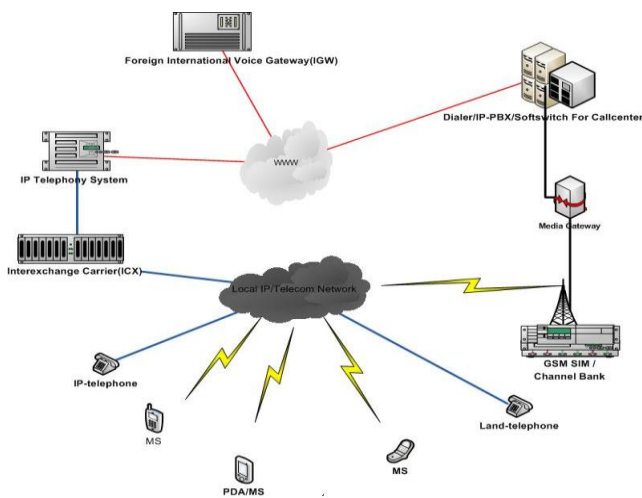


Figure 2 Conceptual diagram of call center infrastructure use for illegal VoIP

Figure 2 shows how illegal VoIP could be operated using the call center infrastructure and transmission facility. If the Multi Router Traffic Grapher (MRTG) of called data is being reviewed, it will only represent the incoming route through call center agents that would be treated as inbound call from foreign customer care services. However, if the call detail record (CDR) is examined from the call center dialer/ IPPBX end, it will show that the originating ID is from outside telecom network and the destination ID is for

local telecom network which clearly indicates the grey call termination using the call center IP [2-3], [5-9], [21-23].

D. Organization of the Article

This article is organized as follows. In introduction section we have stated the situations of call center and a brief literature review on VoIP in Bangladesh. In Section 2, objective of the paper, some proposals regarding technical architectures are described. In Section 3, we have figured out some technical sketches with transmission and core network planning of proposed designs or schemes of call centers. In the section 4 we describe a test bed scenario and validate proposed solution. We have summarized our proposal, the achievement and future indication of development in section 5.

■2.0 OBJECTIVE, PROPOSAL AND METHODOLOGY

Previous researcher had shown the efficiency of call routing and answering machine detection schemes, logically and physically segmented the international and domestic routes (both inbound and outbound) within same system cloud, efficient NGN based switching & transmission (media portability e.g. packet, circuit, IPLC) planning on Call center. Research also progressed in different vulnerability issue with VoIP, Security measurement and smart protection policy & technology for preventing illegal VoIP. But there are very few research on domestic/international call center and illegal VoIP threats because only Bangladesh and India have telecom policy barrier regarding illegal VoIP and Call center [1], [4],[6-9].

In this section the objective of paper, proposals regarding policy amendment, technical, security measurements, nationwide network/gateway monitoring policy and some proposed diagrams for call centers are described.

A. Objective

The main objective of this article is to set a guideline and some technical measurements for commissioning international and domestic call center in same premise or same server cluster avoiding the illegal VoIP. Telecom controlling and regulatory authority (e.g. BTRC (Bangladesh), Ofcom (UK), FCC (USA)) could be amendment their voice based BPO constitutions and take needful steps as per the article [11-14].

B. Proposed Scheme for Telecom Regulator

The Telecom and information authority needs to take the following steps:

- I. BTRC should fix security measure for VoIP issue and Monitoring cell for both technical and legal issues.
- II. BTRC should put restriction only to use SIP (5060-5059) port call center inbound or outbound voice transmission because some UDP port is better than SIP. However, IIG allow from there National Gateway. And network-monitoring authority should inspect on other that port transmission such as Inter Asterisk Exchange (IAX/IAX2). IAX uses port 4569 UDP outbound and inbound for communication [4], [11].
- III. Real time Call Detail Record (CDR) monitoring (from Remote) and mediation (for future Reference) facility have to be developed. So BTRC and other network operation center can be monitored from their Network Operation Center (NOC).

- IV. Real-time monitoring facility for both TDM and SIP trunk (Domestic and International) has to commission and monitoring call center in regular basis [14], [15].
- V. BTRC or other authority can design a system for deep packet inspection (DPI) facility for monitoring call center transmission packet without SIP enable IP address [21]. Normally ISP provides an IP block with one SIP (5060) enabled IP. And the DPI scheme could be applied for voice port blocked IP.

C. Proposed Scheme for Call Center Operators

Some of proposed schemes for call center license holder have to follow:

- i. Call center operator has to properly maintain technical regulation of Call center policy 2009 and Bangladesh-ILDTS Policy 2010 like, taken internet connectivity or private leased line from one ISP or the call center operation and management network should be different.
- ii. Call center operator will clarify the CDR (both Domestic and International), Multi Router Traffic Grapher (MRTG) of usage bandwidth, NMS and other monitoring issue to BTRC. call center(CC) operator will provide the highest level of system access.
- iii. Call center operator need to redesign their call center as per following (Section III) proposed diagrams, which meet the requirement of the main objective.

D. Call Center Re-architectures Proposal

Call center service providers need to reshape their system design as follows:

- i. Physically segment the transmission pipe for domestic and international domain.
- ii. Using same server cloud and network to minimize the cost but highest level of system access and control has to preserve only the monitoring authority and commission.
- iii. Logically routing separation by password protected provision and switching policy system.

3.0 PROPOSED ARCHITECTURES FOR CALL CENTER

This section contains the detail explanation about the above mentioned schemes and technical blueprints.

A. Domestic and International in Physical Transmission Segmentation Model

The first scheme (Figure 3) is to segment the transmission pipe of domestic and international call center and also to separate the provision server for CDR mediation. In domestic call center, time division multiplexing (TDM) communication system is used because calls are routed from PSTN, GSM and CDMA operators. Usually the telecom operator offers ISDN for call center provider. Because most of the telecom use conventional mobile switching center (MSC) and TDM based tandem. The telecom operators usually not convert the signaling from TDM to SIP or H.323. The domestic call center providers supposed to convert the EDSS1 or SS7 to SIP or H.323 using media gateway, E1 PRI card (Sangoma, Digium) [14], [15].

On the other hand, the international call center dialer or PBX application connected to foreign voice carrier or IGW over SIP, IAX, IAX2 or H.323 communication via internet. Figure 3 states

that the domestic and international call centers are separated by transmission line with the separate CDR and dialer. Local area network (LAN) is also separate for both international and domestic service. International and domestic call center, routes the voice using same ISP but the transmission pipe should be physically separate. The domestic call center can be using ISDN-PRI transmission from PSTN & GSM, SIM & channel bank or local IPTSP provided SIP/IAX2 transmission (Via internet). The CDR mediation server is separated and the monitoring authorities can easily store, monitoring the real-time data and can able to find the any domestic /international route mixture or overlapping.

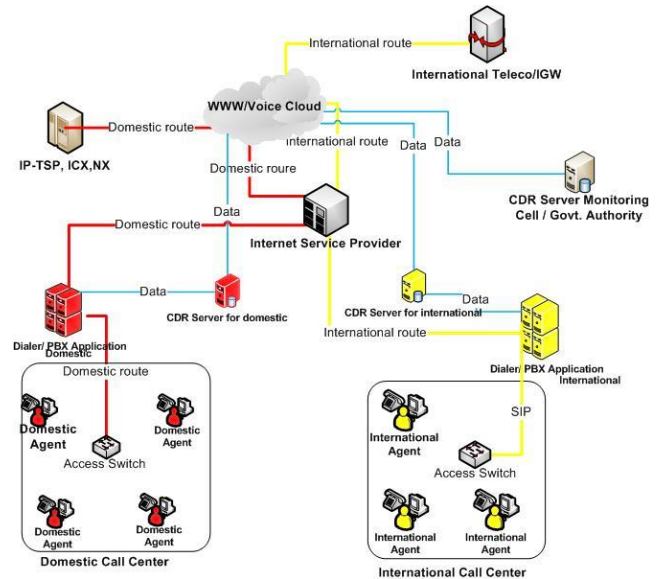


Figure 3 Proposed international and domestic in same call center premise

In such case, the dialer can communicate with foreign voice carrier or minute provider using TCP port 5060-5069 (SIP) or 4569 (IAX). Along with the transmission separation, Call detail record (CDR) mediation has to separate both for international and domestic dialer. Both CDR mediation servers should be connected to internet and both of the servers should act as Web interpreter server. In that case apache or windows based web service need to be enabled for this type of service. So the Government and other monitoring authorities can able to continuously monitoring through HTTP port. So that technical approach can clearly identify any illegal VoIP issue with call centers [4], [15-18].

Currently, in Bangladesh there are many Internet Protocol Telephone Services Provider (IPTSP) are operating in the IP telephony business. Mainly they provide corporate PBX and IP telephony solution. The call center industries are interested about taking their IP telephony services for the domestic call center purposes from late 2010. The reason is that these IPTSP can able to publish non geographical number (NGN) or direct inward dialing (DID). Moreover, the call center providers have found an easy solution for domestic inbound and out bound cases. Because no upfront cost involves in this system like TDM based solution. No need to deploy any additional devices; only need an internet connection or data connectivity from ISP. Fortunately as per BTRC regulation all IPTSP licensee has to have an ISP license. They route the voice to call center through Inter Exchange Carrier (ICX) [2-4], [16-22].

B. Premise Based Domestic and International in Same Server Cluster with Centralize CDR Local Server Model

The other option (Figure 4) could be effective where a central CDR mediation server and a Dialer or private branch exchange (PBX) cluster used. This system can able to provide facility to use single server room, less transmission complexity, unified communication and NGN compatibility, less or no hardware used, robust routing & switching policy, smart monitoring facility & route flow control and modest CAPEX/OPEX model for call center service provider. And there needs to apply a secured authentication policies for the databases and only the government authority and telecom/call center commission have preserved the right to change system policy [24-26].

The server must be connected to the Internet so the authority can able to monitor the system and CDRs day/monthly basis or any time [16], [18-20], [23], [24]. And the call application system, dialer or PBX can be server cloud based and both international and domestic route. The cloud server has to design such a way that the call manager (CM) or voice gateway has to operate by separate IP or transmissions and the database, web (Apache), firewall (both for SIP and HTTP) and routing/switching policy (softswitch) service must be centralized for all call managers and SIP gateways (please see Figure 5). The main task of the softswitch is to provide the routing policy, dial and inbound call forwarding policy, controlling the automatic call distribution (ACD) system, IVR (both for inbound and outbound) and balanced switch policy to route the calls to proper agents.

In this proposed architecture, the agents are connected in same IP subnet. Such as suppose the 192.168.1.X subnet is used where IPs from 192.168.1.2 to 192.168.1.10 is allocated for the server cluster including SIP router, data router, firewall, CM/SIP gateways, soft-switch and database/web services. And IP range have allocated from 192.168.1.11 to 192.168.1.110 for international, 192.168.1.111 to 192.168.1.210 for domestic purposes (both for inbound and outbound for each segments).

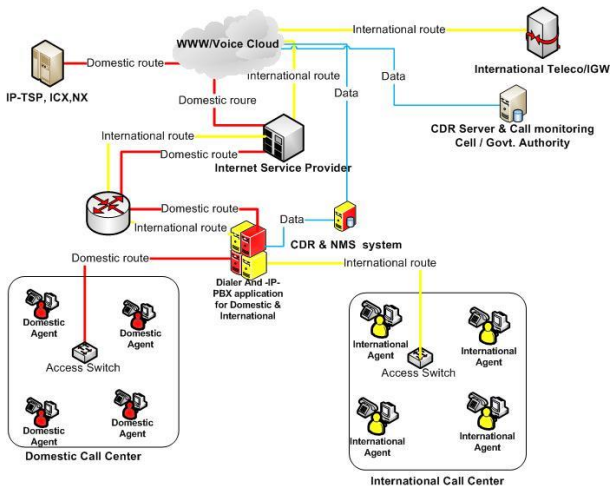


Figure 4 Proposed international and domestic in same call center (CC) premise with central CDR local server with segmented domestic and international route

In Figure 5, we detail out proposed protected core routing and switching processes to distinguish international and domestic calls which has been presented in Figure 4. In this planning, a password-protected policy is introduced. Figure 5a, is a schematic approach of international and domestic dialplan where the configuration files are encrypted at the Call Center Server Cluster. Only the telecom

regulatory and/or call center monitoring authority preserve the password or key of these files. And it is not possible for a call center operator to change any variable. The password, public and private keys are owned by the call center governing authority or Telco/ICT commission. The transmission protocol is SIP or IAX for both international and domestic route. The all level of data and voice gateway based routing policy must be used secured IPsec and private/public keys [16-20], [22-26].

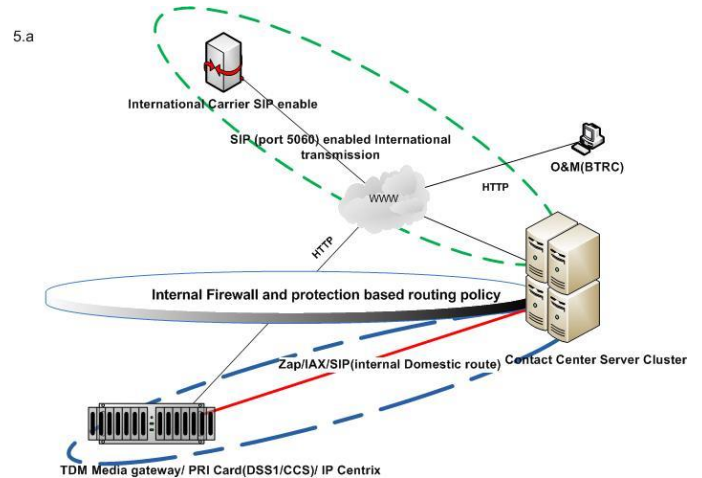


Figure 5 (a) Core routing & switching planning (international & domestic)

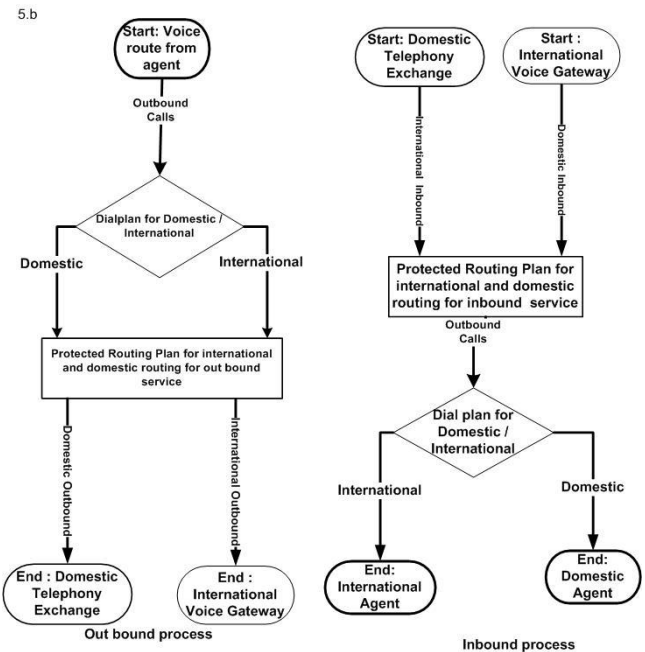


Figure 5 (b) Inbound and outbound call flow processes for secured call center cluster

In the routing technique, international and domestic calls are separated which is described in the dialplan. In the switching, an additional script exists to protect the dialplan from other hidden dialplans or unethical interventions e.g., SIP attack, eavesdropping, etc. Call flow processes (i.e., inbound and outbound) are presented in diagram Figure 5b. We have separated the flow chat in two steps for inbound and outbound process for simplicity. Both processes

have same procedure step but inbound is bottom up approach and outbound is top-down method:

The step-by-step algorithm to reduce illegal VoIP calls is illustrated below:

- Step 1: Call Initiation of extensions for both domestic and international agents. Here destination address (domestic/international), non-geographical number selection (for inbound/outbound and domestic/international calls), and transmission parameters are defined.
- Step 2: Match and patch domestic/international route with dialplan and with call center agents. The matching is done using extensions.conf and sip.conf files. An additional script is added to deactivate hidden extension file(s) and make the dialplan files as read-only and non-changeable. Figure 6 gives a detail dialplan scripting.
- Step 3: Hereafter a routing plan is deployed to separate the domestic and international routes.

Most of the call center application, dialer or soft PBX customized from some well-known open source Linux OS based application named Asterisk, FreePBX, FreeSwitch etc. And all above mentioned system has different configuration files or templates to create voice routing, agent extension configuration, ACD or IVR. The configuration file or template could be command or graphical based. There should also be a provision for putting password on these systems to make very safe and secured policy for avoiding the illegal voice termination by the call center provider or anybody [16 -19], [22-26].

4.0 TEST-BED SENARIO AND RESULT ANALYSIS

A. Background of the Test Bed

After the regulation to setup international and domestic calling system in same premise but different horizontal states, BTRC informally asked to create a test bed or run a short term Pilot project based on proposed architectures and scheme and BTRC also agreed to provide the facility for temporary NMS, DPI and test CDR provision for specific test bed or pilot. As per BTRC’s instruction the first phase has to be conducted by two week and next phase (means in same server and transmission line) should be conducted by one week only. Virgo Contact center Service Ltd (one of the largest Call center in Bangladesh) was selected for test bed for international and domestic operation in same premise but different horizontal. That time Virgo was dealing with one international (overseas GP healthcare/telemedicine provider, UK based) and domestic medical call center (local hospital telemedicine consultation) projects.

B. Implementation of Pilot Test Bed

Phase I:

First we have applied the test scenario for transmission and CDR mediation segmentation (For detail please see Section III.A and Figure 3) model. Instead of deploying the time division multiplexing (TDM-ISDN PRI) transmission for domestic call center purposes, we have selected a local Internet Protocol Telephony Service Provider (IP-TSP). Thus the complexity, reliability and security have become very challenging. Because we were also using a foreign IGW for international voice transmission and both international and domestic routes are transmitted though same transmission pipe or same ISP thought the IP block and route plan was separated. And the CDR mediation application, predictive

dialer and local network for international and domestic call centers were also separated. We have also built a local network monitoring system (NMS) with the one of the open source solution for local and domestic traffic monitoring and a FTP and apache (Web) for CDR monitoring, mediation and provisioning. On the BTRC monitoring cell side, their technical team was also monitoring the real time CDR and voice traffic for both foreign and local. The phase I testing period was lasting for 15 day and came out a successful result and satisfaction of BTRC.

Phase II:

After the successful experimentation of Phase I, test for Phase II has been prepared. The test includes a cluster dialer application, CDR server, single switching & routing field (please see Figure 5), single SIP gateway with security and same international and domestic call routing policy that are physically local network but logically separated (For detail please see Sec III.B and Figure 4).

In routing policy state, we route the international call through BTRC allocated SIP port (5060-5061) open IP [8] and route the domestic call to another IP in same block to connect with local IPTSP. Here the local ISP has its own IPTSP solution and connected to inter carrier exchange (ICX). The system had to built very carefully and gave all technical authority to BTRC for checking the routes, traffic, CDR and only they can able to operate the policy state.

```

SIP configuration class and file
[Local_IPTSP]
username=XXXX
type=friend
secret=XXXX
host=XXX.XXX.XXX.XXX
dialplan=call
allow=729d9dmg711du1awdalan
canreinvite=yes
insecure=early
context=from-trunk

[International_IPTSP]
username=XXXX
type=peer
secret=XXXX
host=XXX.XXX.XXX.XXX
dialplan=call
allow=729d9dmg711du1awdalan
canreinvite=no
insecure=port,invite
context=from-trunk

[from-trunk-sip-Local_IPTSP]
include => from-trunk-sip-Local_IPTSP-custom
exten => _.,1,Set(GROUP)=OUT_1
exten => _.,n,goto(from-trunk,${EXTEN},1)

[from-trunk-sip-International_IPTSP]
include => from-trunk-sip-International_IPTSP
exten => _.,1,Set(GROUP)=OUT_2
exten => _.,n,goto(from-trunk,${EXTEN},1)

; end of [from-trunk-sip-Local_IPTSP]
; end of [from-trunk-sip-International_IPTSP]

Main Dial Plan Macro, class and file
[outbound-allroutes]
include => outbound-allroutes-custom
include => outrt-1; Local_IPTSP
include => outrt-2; International_IPTSP
exten => foo,1,noop(bar)

[outrt-1]; Local_IPTSP
include => outrt-1-custom
exten => _01-[5-9]XXXXXXXXX,1,Macro(user-callerid,SKIPPTL)
exten => _01-[5-9]XXXXXXXXX,n,noop(calling out route: bicon)
exten => _01-[5-9]XXXXXXXXX,n,Macro(record-enable,${AMPUSER},OUT_)
exten => _01-[5-9]XXXXXXXXX,n,Macro(dialout-trunk,2,${EXTEN},)
exten => _01-[5-9]XXXXXXXXX,n,Macro(outisbusy,)
exten => _02-XXXXXXXX,1,Macro(user-callerid,SKIPPTL)
exten => _02-XXXXXXXX,n,noop(calling out route: Local_IPTSP)
exten => _02-XXXXXXXX,n,Macro(record-enable,${AMPUSER},OUT_)
exten => _02-XXXXXXXX,n,Macro(dialout-trunk,2,${EXTEN},)
exten => _02-XXXXXXXX,n,Macro(outisbusy,)

; end of [outrt-1]

[outrt-2]; International_IPTSP
include => outrt-2-custom
exten => _0044XXXXXXXXX,1,Macro(user-callerid,SKIPPTL)
exten => _0044XXXXXXXXX,n,noop(calling out route: International_IPTSP)
exten => _0044XXXXXXXXX,n,Set(WORKCLASS=${if($WORKCLASS=""?default:${WORKCLASS}})
exten => _0044XXXXXXXXX,n,Macro(record-enable,${AMPUSER},OUT_)
exten => _0044XXXXXXXXX,n,Macro(dialout-trunk,4,${EXTEN},)
exten => _0044XXXXXXXXX,n,Macro(outisbusy,)

; end of [outrt-2]

This dial plan clearly indicate that there is no mixture of international and foreign route and for security reason different class, macros and multi dimension file were used

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Figure 6 SIP configurations, route plan and dial policy for international & local route

We have developed a cluster for dialer, soft switching, IP-PBX system based on asterisk 1.8.7 (open source voice solutions) and a secure SIP routing policy algorithm and dial plan as displayed in Figure 6. After final acceptance test (FAT) of phase II, the voice routing and cluster application for international & domestic run smoothly. The test period was seven days.

A. Phase I and Phase II Test Bed Results and Analysis

For Phase I and Phase II we demonstrate some statistical result based on real time CDR, MRTG and system generated log from both test beds.

After successful completion of Phase I and Phase II, we have formulated two real-time graphs from day to day CDR and system log files (please see Figure 7 and Figure 8). Most interestingly, the ratio of the two components has a profound effect on the microscopic structure and macroscopic properties of the gel in toluene.

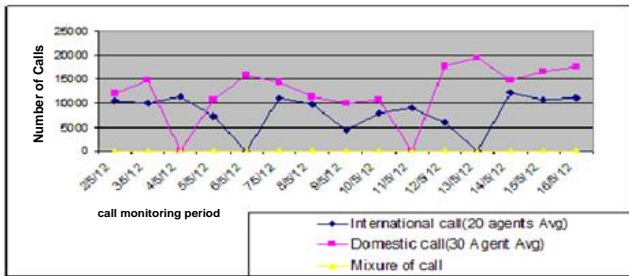


Figure 7 CDR presentation for phase I for pilot test bed

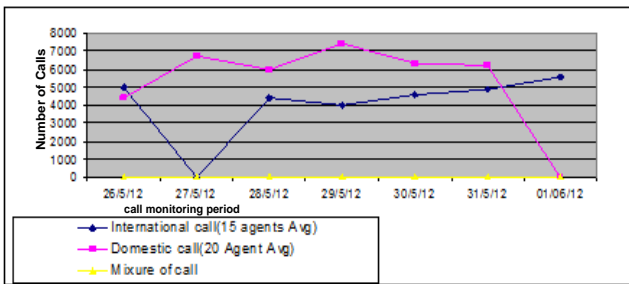


Figure 8 CDR presentation for phase II for pilot test bed

Figure 7 shows the number of international, domestic and calls mixture (which indicates the illegal call termination or origination phenomena). And it was found there was no call mixture. And the call pattern of international (20 agents) and domestic (30 agents) and ratio were more less have same in each day.

There was no call during weekends (for local Friday and international (UK) Sunday) both for Phase I and Phase II. So the phases I, there are no calls on 4th & 11th may 2012 no calls for domestic calls and 6th & 13th may 2012 for international and in phase II, call quantity is void on 26/5/2012(international) and 01/06/2012 (domestic).

The average latency within the pilot test period (Phase I & II) of the foreign and domestic is described in Figure 9 and 10. In both test phases we have used one of the UK gateway for international testing and a Bangladeshi IP-TSP for domestic call testing. Normal latency for UK IGW is 450 ms to 500 ms and on other hand Bangladesh IP-TSP latency is very low (e.g. 20 ms to 25 ms). In Figure 9, the graph represents phase I test bed average latency state of both domestic and international route.

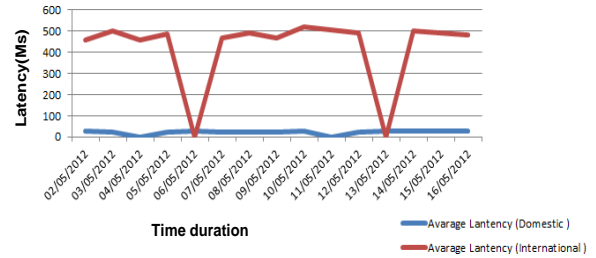


Figure 9 Average Latency (ms) period for domestic and international call center in Phase I

The data is retrieved from system log form both international and domestic dialer. The overall average latency (for phase I, two weeks) for domestic and international is 25.66ms and 485.25ms. On May 04, 06, 11 and 13, 2012 no calls on system and the latency is zero. The average latency graphical presentation for Phase II is illustrates in Figure 10. In Phase II, the total average latency is 27.75ms for domestic route and 491.18ms for international route and there is no call on May 27 (International) and June 1 (domestic) because of day off or weekend.

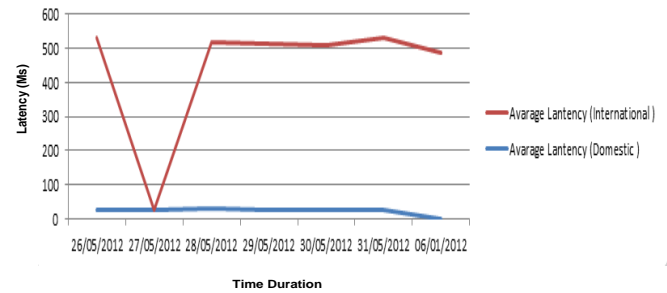


Figure 10 Average latency (ms) period for domestic and international call center in Phase II

From Figure 9 and Figure 10, it is clearly visible that the international and domestic route’s latency is static and linear manner. It means that international route not mixed with domestic route. Actually we have calculating that routing parameter like latency from route end to agent or other end extension. And circuit and routing hop and destination is fixed. So if any illegal VoIP happened we have seen some abnormal shape of graphs like sudden spikes. So from the Figure 9 and Figure 10, it is very much understood that the proposed solutions and routing policy working properly and VoIP reduction scheme working. The latency could be fluctuating if the transmission (both inside and outside of country) is bad.

BTRC NMS unit also confirm the same call ratio and they also differential the routes for foreign and domestic. On the other hand, despite the fact that we also get the same pattern of call from Figure 8 and no mixture call, BTRC was fully not sure about the system and they want time to understand the route algorithm policy, security issue and the modules and macros that we have used in Phase II. Also they are not able to fully monitoring the routes because they don’t have any provision for Packet inspection mechanism such as DPI also they don’t have fully control of attached ISP, IPTPS, ICX and border Gateway for foreign international gateway(IGW). So the second technical scheme or phase II is still under consideration for authority’s approval. But from technical point of view, phase I and II the duration that allocated by BTRC it is enough to verify the proposed system designs of both international, domestic services within one server

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