



# **Call Center Suite**

## **Proposal**

Prepared for

# **Indus Hospital**

Prepared by



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# 1. Executive Summary

This proposal has been prepared by **SyncTech Solutions** [Hereafter **SyncTech**] for **Indus Hospital** [Hereafter **Indus**] to provide an Asterisk base call center suite. It will also illustrate list of features of SyncTech's state-of-Art call center solution.

With our cumulative experience of over a decade in Telecommunication, Voice over IP Technology and networking infrastructure expertise, we are aware of the technological challenges. Our offerings are tailored so that these challenges are met very effectively and you will overcome the challenges that are currently being faced.

## 2. SyncTech Profile

### 2.1. Introduction

SyncTech Solutions provides business software and solutions customized for your business. Whether it is service industry or manufacturing products, SyncTech aims to help automate and manage entire business. SyncTech specialized in providing a complete picture and control of business processes like customer demand, forecast, orders, suppliers, inventory, manufacturing, sales, distribution, customer support and customer service. With our software you can manage your finances, Marketing, Operations & Marketing. We specialize in serving small to medium size businesses ready or looking for growth. We help you with tools, technology and best practices in your industry.

SyncTech is a leading company in the field of software designing, web development and voice communication systems in Karachi, Pakistan. We have sound features in that field. Quality of service is our promise to all the clients. Our vast experience in the field of Voice solution can help to boost your business. Additionally, we provide all kind of Business, Industrial, Banking, Medical, Finance, Educational, Accounts and Manufacturing software. We have decided to convert every manual work in the form of software. Now a days communication is an essential part of every business or industry. Reliable and clear voice communication is an easy way to get work or customers from all over the world satisfied. So, we say that it is the time to have an efficient and at the same time cost effective voice communication system for your customers.

### 2.2. SyncTech Services

#### 2.2.1. IT Solutions

##### ❖ VoIP Solutions:

- Calling Card / Painless platform
- Broadband / IP Telephone
- IP-PBX / IVR system
- Call Center solution

##### ❖ Open Source Solutions:

- Linux server services
- Shifting users to Linux GUI environment
- Support with open source software's

❖ **Web service:**

- Web Designing & Development
- E-commerce platform
- Web Applications
- Mobile Applications
- Portals

❖ **Domain & Hosting services**

### **2.2.3. BPO Services:**

❖ **Call Center:**

- Customer service
- Survey
- Appointment setting /scheduling

❖ **Non-Voice:**

- Branded SMS Services
- Back office service
- Data entry

### 2.2.3. Our Clients:



alshaheer



PIRANI GROUP OF COMPANIES



#### 2.2.4. Contact Information:

Company Name	SyncTech Solutions
Address	B-134, Block 2, Gulshan-e-Iqbal, Rashid Minhas Road.
City	Karachi
Country	Pakistan
Phone	+92-300-3497670
E-mail	info@synctechsolutions.com

### 3. Call Centre Suite

Having cumulative experience in open source voice solutions for IP Telephony and CRM Applications, SyncTech can provide solution to best fit industry's needs and at the same time saving initial cost with GPL licensed software and operational cost as well.

SyncTech will deploy an Asterisk base call center solution at **Indus** office which will cover the above mentioned requirements. However; the system is capable of many advance requirements for communication and complete call center suit; it can also be customized for any additional feature required respectively.

Our core focus is to provide you with affordable state of art solution that best fit your need and cutting down your telecommunication expenses while also avoiding heavy licensing costs. Below is a list of some features that Asterisk can be upgraded to.

#### ***Why we think our Call center suite is the solution for you?***

- Highly feature rich
- Matured, reliable & scalable
- Detailed reporting
- Excellent community support
- GPL licensed (No licensing fee)
- Very low procurement & support cost
- Very flexible for customization
- Can easily be integrated with third party software
- Can be remotely operated

***You can rest assured that the most powerful, recognized and stable combination of software is being used creating a solution that is 100% scalable to outgrow your wildest expectations!***

#### **3.1. Asterisk**

Asterisk is a complete PBX system. It runs with Linux and provides the entire basic features that can be expected from a PABX and lot more.

##### **Major Features:**

- Unlimited Extensions
- Mobility (Extensions on mobile)
- Remote Extensions
- Multiple location connectivity
- Unlimited Voicemail
- Audio conferencing
- Call waiting



- Call Recording
- Caller ID (CLI)
- IVR
- Detailed CDR & reports
- Call restrictions on extension or group
- Least Cost routing.

*Detail information of Asterisk and its features are in Annexure*

### **3.2. Call Center suite**

Call Center suite functions as an ACD for inbound calls or for making outbound calls. The proposed solution will handle both Inbound & Outbound calls along with features listed below.

#### **Features:**

##### **Main Features:**

- Inbound, Outbound and Blended call handling
- Automated Call Distribution (ACD)
- Call Telephony Integration (CTI)
- Web-based agent and administrative interfaces
- Call Recording
- IVRs and Voicemail boxes

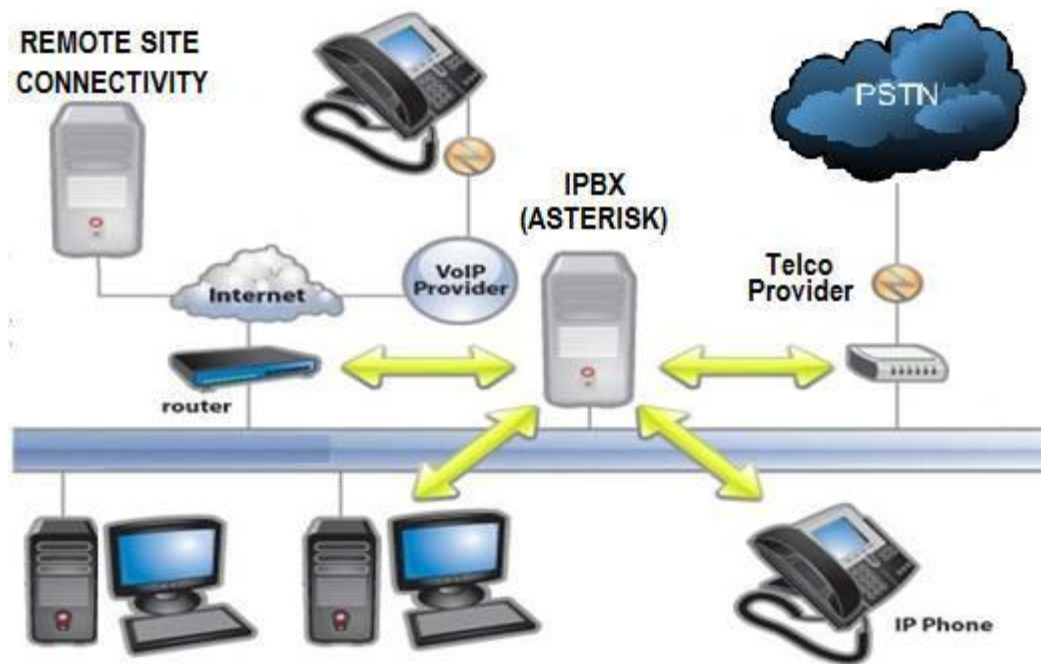
##### **Agent Interface Features**

- Mini-CRM window pop up
- Call Duration Alert
- Call back
- 3 way call conferencing
- Call Transfer
- Call disposition selection

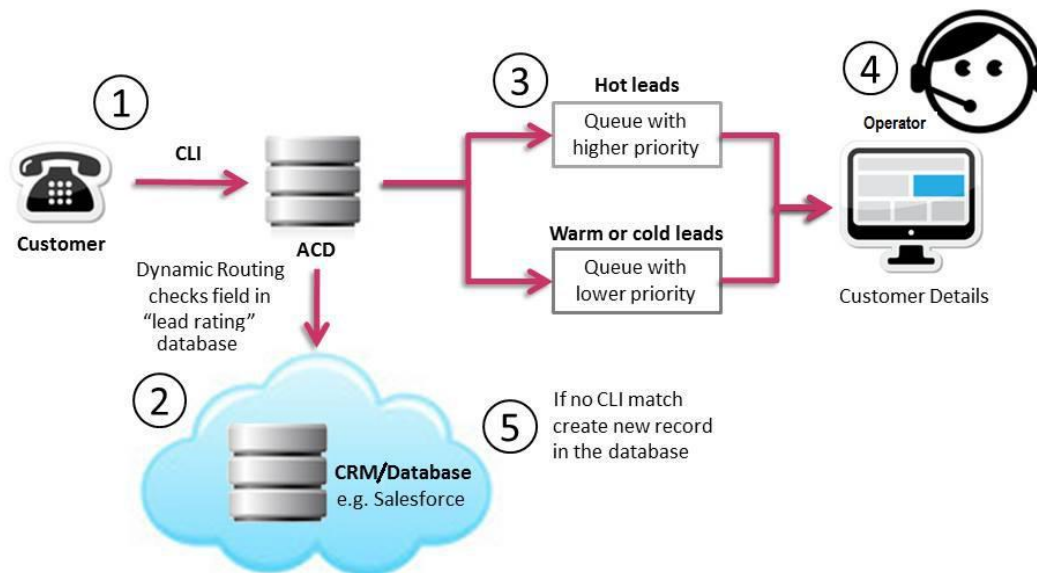
##### **Manager Interface Features**

- Agent statistics
- Live Monitoring
- Call Barging (Call Coaching)
- Wallboard
- Call summary report
- Agent performance report
- Campaign report

### **A brief pictorial walk through**



*This shows how a complete PABX network is laid with a connection to remote Asterisk through voice enabled data connection.*



*This figure shows call distribution, queue priority (if set) and integration with any other database, i.e CRM, ERP, etc*



Smart call routing is a great way to save on your telecommunication cost. System using a set criteria routes the calls through your selected carries or set priorities

Logged in as User: 6666 on Phone: IAX2/cc350 to campaign: TESTCAMP [GROUPS](#) [LOGOUT](#)

**VICIdial** SCRIPT 2009-02-12 02:29:55 session ID: 8600051 Calls in Queue: 0 **NO LIVE CALL**

STATUS: seconds:

**PAUSE** **RESUME** Customer Time:  Channel:

☐ ALT PHONE DIAL Customer Information:

RECORDING FILE: Title:  First:  MI:  Last:

RECORD ID: Address1:

**START RECORDING** Address2:  Address3:

**WEB FORM** City:  State:  PostCode:

**PARK CALL** Province:  Vendor ID:  Gender:

**TRANSFER - CONF** Phone:  DialCode:  Alt. Phone:

**HANGUP CUSTOMER** Show:  Email:

**SEND DTMF**  Comments:

[1 ACTIVE CALLBACKS](#) [ENTER A PAUSE CODE](#)

[MANUAL DIAL](#) [FAST DIAL](#)

VICIDIAL web-client version: 2.0.5-197 BUILD: 90209-0132 Server: 10.0.0.6 **HOT KEYS INACTIVE**

[Show conference call channel information](#) **MUTE**

[Alert is OFF](#)

Agent interface, which shows agents options, i.e pause or ready, placing a call on hold, customer information

## Real-Time Main Report

Choose Report Display Options

RELOAD NOW

MODIFY | SUMMARY

refresh: 31

VIEW LESS

VIEW USER GROUP

SHOW SERVER INFO

HIDE WAITING CALLS

SHOW IN-GROUP STATS

SHOW PHONES

SHOW CUSTPHONES

DIAL LEVEL: 1.000 TRUNK SHORT/FILL: 0 / 0 FILTER: NONE TIME: 2018-09-05 14:25:34  
 MAX LEVEL: 3 DROPPED MAX: 3% TARGET DIFF: 0.0000 INTENSITY: 0  
 DIAL TIMEOUT: 60.0000 TAPER TIME: 2100 LOCAL TIME: 24hours AVAIL ONLY: N  
 DIALABLE LEADS: 15985 CALLS TODAY: 3361 AVG AGENTS: 0.43 DIAL METHOD: RATIO  
 HOPPER (min/auto): 36 / 1 DROPPED / ANSWERED: 48,000 / 946 DL DIFF: 0.27 STATUSES: NEW  
 LEADS IN HOPPER: 22 DROPPED PERCENT: 5.07% DIFF: 63.13% ORDER: DOWN

current active calls 13	calls ringing 11	calls waiting for agents 0	calls in IVR 0	chats waiting for agents 0	callback queue calls 0
agents logged in 27	agents in calls / dials 9 / 1	agents waiting 13	paused agents 3	agents in dead calls 0	agents in dispo 2

## Agents Time On Calls Campaign:

2018-09-05 14:25:34

STATION	USER	SHOW ID	INFO	SESSIONID	STATUS	PAUSE	MM:SS	CAMPAIGN	CALLS	HOLD	IN-GROUP
SIP/529	529 +			8600072	READY		10:27	45816529	1468		
SIP/519	519 +			8600051	READY		0:11	45816519	1408		
SIP/526	526 +			8600061	READY		8:39	45816526	1036		
SIP/528	528 +			8600075	READY		5:20	45816528	1274		
SIP/518	518 +			8600067	READY		7:08	45816518	1015		
SIP/523	523 +			8600069	READY		4:08	45816523	1606		
SIP/515	515 +			8600052	READY		4:19	45816515	1982		
SIP/507	507 +			8600062	READY		1:27	45816507	2073		
SIP/512	512 +			8600073	READY		2:35	45816512	1683		
SIP/516	516 +			8600068	READY		1:36	45816516	1999		
SIP/517	517 +			8600074	READY		1:49	45816517	1747		
SIP/504	504 +			8600053	READY		2:02	45816504	1063		
SIP/505	505 +			8600070	READY		0:50	45816505	2036		
SIP/513	513 +			8600064	INCALL	A	9:57	45816513	1587		
SIP/506	506 +			8600077	INCALL	I	9:42	45816506	1267	0	506 - 506
SIP/511	511 +			8600071	INCALL	A	7:50	45816511	1626		
SIP/509	509 +			8600060	INCALL	A	3:08	45816509	1080		

Live Dashboard will keep you upto dated with what your agents are up to. On call, pause time, number of call in queue and many other campaign details.

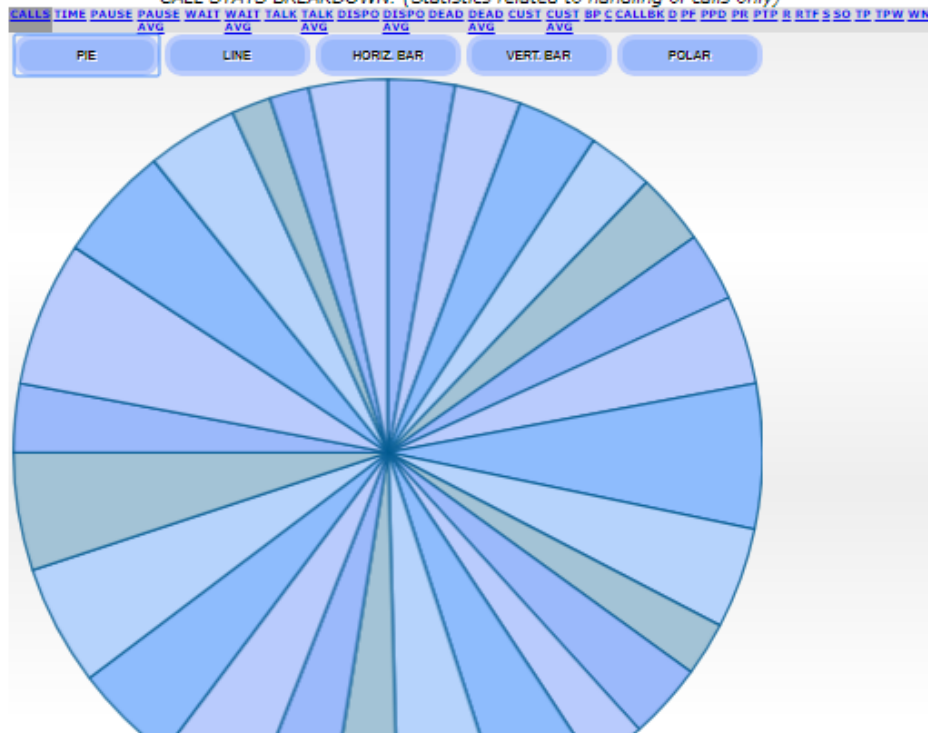
## Agent Performance Detail ?

<b>Dates:</b> 2018-09-05 00:00:00 to 2018-09-05 23:59:59	<b>Campaigns:</b> -- ALL CAMPAIGNS -- 45818500 45818501 45818502 45818503	<b>User Groups:</b> -- ALL USER GROUPS -- ADMIN CPAgent	<b>Users:</b> -- ALL USERS -- 500 - 500 501 - 501 502 - 502 503 - 503	<input type="checkbox"/> Only live agents <input type="checkbox"/> Show %s <input type="checkbox"/> Time in seconds <input type="checkbox"/> Show date breakdown <input type="checkbox"/> Search archived data <input type="checkbox"/> Show defunct users	<b>Display as:</b> HTML Shift: --	<a href="#">REPORTS</a> <input type="button" value="SUBMIT"/>
---	--	--	--	---	--	--

Agent Performance Detail 2018-09-05 14:26:28  
 Time range: 2018-09-05 00:00:00 to 2018-09-05 23:59:59

----- AGENTS Details -----

### CALL STATS BREAKDOWN: (Statistics related to handling of calls only)





This is the main area where all types of reports and server details can be accessed from.

## 4. Proposed Solutions

SyncTech will deploy an Asterisk based call center suite for **Indus**.

## 5. Hardware Requirements

SyncTech expect its clients to provide them with the following hardware and software prior to deployment of our solutions.

- Server – Intel® Xeon® 3.2 GHz Quad core with 8 GB Ram and 1 TB SATA Hard
- Headphones
- Client PC – Core i3 2<sup>nd</sup> generation with 4 GB RAM
- Static Public IP(for server remote access)

## 6. Quotation – Asterisk PBX configuration

### Service Charges

Description	Rate (Rs.)
Call center deployment <ul style="list-style-type: none"><li>Linux OS installation</li><li>Asterisk Installation &amp; Configuration</li><li>Call center suite Installation &amp; configuration</li><li>Trunk configuration</li><li>User Training</li><li>Integration with CRM</li></ul>	195000
<b>Total Service Charges in PKR</b>	<b>195000</b>
<i>*Rate quoted above are exclusive of all taxes, taxes will be add as per FBR;s SRO.</i>	
<i>*Above quotation rate is as per discussion any changes or any customization may be charged separately.</i>	
<i>*In case of any license module required client will pay that directly to the vendor.</i>	
<i>*In case of any traveling required outside of Karachi, client will bare travel and lodging expenses.</i>	

### Payment Terms

60% in advance

40% after completion and final testing

Hardware charges (if any) are to be paid in complete advance.

Payments are accepted in Pakistani Rupees only in shape of cheque or pay order.

Cheques should be made in favor of **SyncTech Solutions**.

### Duration:

Complete Installation & configuration may take up to 15 working days after procurement of complete hardware requirement.

### Training:

Administration training will be provided.

### System Support:

One month of system support will be provided free of cost.

## 7. Installation & Configuration

After procurement of hardware and other requirements, SyncTech engineers will setup the applications and establish a connection between Asterisk and trunk line. This would enable all **Indus** system to seamlessly access the server and send/receive calls via trunk lines. Testing and enabling of the calls would also be taken care of at this time and any problems will be addressed. Installation and Configuration will be done remotely and/or on-site by SyncTech engineers.

Server setup will include the following;

- Installation from scratch (OS & Asterisk)
- Trunk Configuration
- Customization (if any)
- QC and Testing
- Training

A final quality control check would ensure:

- Minimal or no line echo testing.
- Acoustic echo correction.
- System load testing using automatic call generators.



## 8. System Support

Support is accessible by calling our account managers and even by sending an email to support@synctechsolutions.com an automatic ticketing system will open a ticket in your name which can be tracked for its entire life cycle. The tickets are view-able over the web by **Indus's** personnel at any time.

### Support

After free support, SyncTech can provide extended support by signing and SLA with the client.

# Annexure – Introduction to Asterisk™

Asterisk is a complete PBX in software. It runs on Linux and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in three protocols, and can inter-operate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Check the Features section for a more complete list.

Asterisk needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium™. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one to four-port modular FXS and FXO card.

Also supported are the Internet Line Jack and Internet Phone Jack products from Quicknet.

Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces. Asterisk supports US and European standard signaling types used in standard business phone systems, allowing it to bridge between next generation voice-data integrated networks and existing infrastructure. Asterisk not only supports traditional phone equipment, it enhances them with additional capabilities.

Using the Inter-Asterisk eXchange (IAX™) Voice over IP protocol, Asterisk merges voice and data traffic seamlessly across disparate networks. While using Packet Voice, it is possible to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

Asterisk provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

## IP Phone Integration

Any SIP based phone integrates to Asterisk. SyncTech will provide a list of choices for IP Phones.

## Asterisk™ Features

Asterisk is all encompassing in its features such that it provides for most telephony functions within without the need to install third party software or hardware. Some of the features Asterisk provisions for include:

- Unlimited Voicemail
- Multiparty Conferencing
- Following Caller-id and Call transfer
- Sip Video
- Call Forwarding
- Call Recording
- Built in fully programmable IVR
- Music on Hold and transfer
- Remote Call Pickup

Asterisk-based telephony solutions offer a rich and flexible feature set. Asterisk offers both classical PBX functionality and advanced features, and inter operates with traditional standards-based telephony systems and Voice over IP systems. Asterisk offers the features one would expect of a large proprietary PBX system such as Voice mail, Conference Bridging, Call Queuing, and Call Detail Records. Some additional features are as follows:

## Call Features

- Automated Attendant
- Blacklists
- Blind Transfer
- Call Detail Records
- Call Forward on Busy
- Call Forward on No Answer
- Call Monitoring
- Call Parking
- Call Queuing
- Call Recording
- Call Routing (DID & ANI)
- Call Transfer
- Call Waiting
- Caller ID
- Caller ID Blocking
- Caller ID on Call Waiting
- Conference Bridging
- Interactive Voice Response (IVR)
- Local and Remote Call Agents
- Music On Hold
- Music On Transfer
- Volume Control
- Remote Call Pickup
- Route by Caller ID
- Time and Date
- Trunking
- VoIP Gateways
- Voicemail
- Computer-Telephony Integration
- AGI (Asterisk Gateway Interface)
- Graphical Call Manager

- Predictive Dialer

## Scalability

- TDMoE (Time Division Multiplex over Ethernet)
- Allows direct connection of Asterisk PBX
- Zero latency
- Voice-over IP

## Codecs

- G.711 (A-Law &  $\mu$ -Law)
- G.723.1 (pass through)
- G.729 (through purchase of commercial license through Digium)
- GSM

## Protocols

- IAX™ (Inter-Asterisk Exchange)
- SIP (Session Initiation Protocol)

## Traditional Telephony Interoperability

- FXS
- FXO