

SANOG XXI

Call Center, Quick Start

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Session Goal



To provide you a brief idea about ip contact center. and guide you to setup your first very basic ip contact center with asterisk.

Agenda



- What is Call Center? (10%)
- Why you need a Call Center Application? (25%)
- Setting Up, The easy way. (5%)
- Know what you are doing, build it yourself. (60%)

What is Call Center?



"A call center or contact center is a centralised office used for the purpose of receiving or transmitting a large volume of requests by telephone"

- wikipedia



What is Call Center?



Types of Call Centers:

Inbound:

"An inbound call centre is operated by a company to administer incoming product support or information inquiries from consumers"

Outbound:

"Outbound call centers are operated for telemarketing, solicitation of charitable or political donations and debt collection."

Agenda

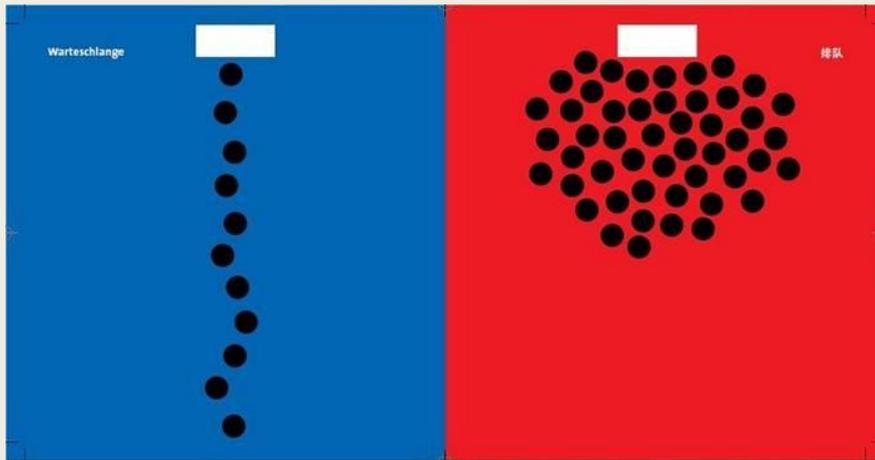


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Why Call Center App?



You need a call center app to provide your service smooth by keeping client happy and in line.



Why Call Center App?



Beside managing the queue it helps..

- Improving Employee Efficiency
 - includes agents proficiency tracking and distributing calls to appropriate agents, etc.
- Refining Customer Service
 - includes giving customer a custom service through taking calls to appropriate agents and giving agents detail idea about the customer through CTI etc..
- Reporting Metrics for Management
 - multiple report to management including call volume, support timing need of access agents or additional trunk etc.

Why Call Center App?



Boosting Performance

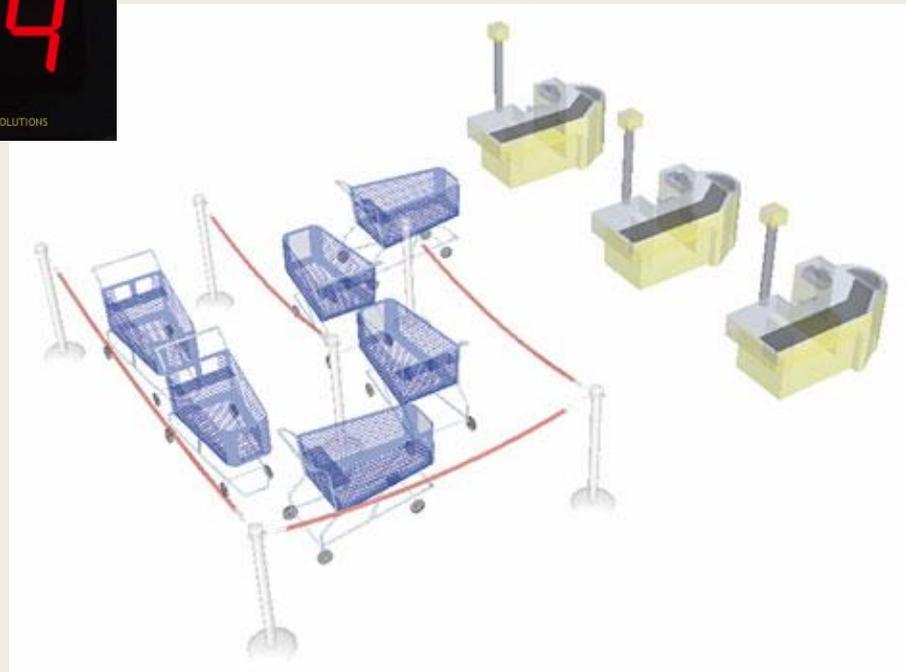
There is lots of way a call center application can boost your service performance. But there is few very important area we should zoom in.

- Inbound:
 - ACD (Automatic Call Distributor)
 - CTI (Computer Telephony Integration)
 - SBR (Skill Based Routing)
- Outbound
 - Natural Predictive Dialing
 - CPA (Call Progress Analysis)

Why Call Center App?



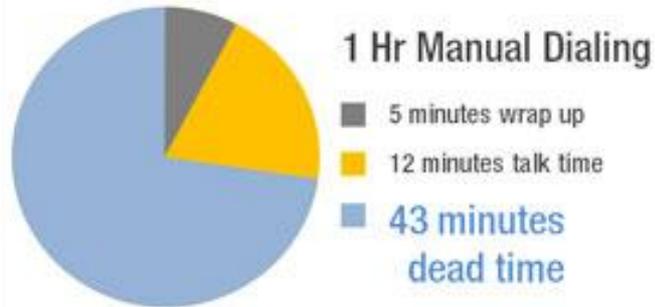
Inbound: ACD, SBR, CTI



Why Call Center App?



Outbound: PD, CPA



Agenda



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Setting Up, The easy way.



Setting up a call center is easy.

just download one of the most used opensource call center application ISO and install it on you hardware.

its only matter what to choose. just google "open source call center software". you get vicidial on 1.

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

VICdial 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
RECORDING FILE:

RECORD ID: **START RECORDING**

WEB FORM

PARK CALL

TRANSFER - CONF

HANGUP CUSTOMER

SEND DTMF

Customer Information:

Title: First: MI: Last:

Address1:

Address2: Address3:

City: State: PostCode:

Province: Vendor ID: Gender: U - Undefined

Phone: DialCode: Alt. Phone:

Show: Email:

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](#)

Setting Up, The easy way.



Just download the ISO, Burn it to CD and boom....
you will get more than anything you need.

VICIdial Users Campaigns Lists Scripts Filters In.Groups User.Groups Remote Agents Admin Reports

VICIDIAL Real-Time **Choose Report Display Options** STOP SLOW GO MODIFY SUMMARY
 DIAL LEVEL: 2.088 TRUNK SHORT/FILL: 0 / 0 FILTER: NONE TIME: 2009-03-30 23:11:16
 DIALABLE LEADS: 16590 CALLS TODAY: 73406 AVG AGENTS: 4.56 DIAL METHOD: ADAPT_TAPERED
 HOPPER LEVEL: 500 DROPPED / ANSWERED: 906 / 10281 DL DIFF: 0.34 STATUSES: NA, A, DROP, N, A6, B
 LEADS IN HOPPER: 540 DROPPED PERCENT: 8.81% DIFF: 7.46% ORDER: DOWN COUNT 4th NEW
 CONTACTS: 6230

- VIEW MORE HIDE USER GROUP SHOW AGENT NAME SHOW SERVER INFO HIDE WAITING CALLS SHOW PHONES HIDE CUSTOMERS

19 current active calls 11 calls ringing 2 calls waiting for agents 0 calls in IVR
 23 agents logged in 15 agents in calls 5 agents waiting 2 paused agents 1 agents in dead calls

VICIDIAL: Calls Waiting 2009-03-30 23:11:16

STATUS	CAMPAIGN	PHONE NUMBER	SERVER_IP	DIALTIME	CALL TYPE	PRIORITY
LIVE	INBOUND	310551646	192.168.1.100	0:23	IN	4
LIVE	INBOUND	310553480	192.168.1.100	0:14	IN	4

VICIDIAL: Agents Time On Calls Campaign: (TESTCAMP) 2009-03-30 23:11:16

STATION	USER	USER GROUP	SESSIONID	STATUS	COST PHONE	HL SS	CAMPAIGN	CALLS
Sep/14-1	1102	ROCPA	8600068	READY		0:14	TESTCAMP	428
Sep/21-1	2139	ROCPA	8600067	READY		0:03	TESTCAMP	618
Sep/18-1	2111	ROCPA	8600054	READY		0:14	TESTCAMP	519
Sep/1-1	2331	ROCPA	8600061	READY		0:15	TESTCAMP	304
Sep/15-1	1099	ROCPA	8600057	READY		0:04	TESTCAMP	504
Sep/9-1	1741	ROCPA	8600055	INCALL A	6615559038	5:01	TESTCAMP	434
Sep/4-1	1159	ROCPA	8600072	INCALL I	2095553064	3:39	TESTCAMP	48
Sep/5-1	2029	ROCPA	8600064	INCALL I	4615552197	3:08	TESTCAMP	84
Sep/6-1	1625	ROCPA	8600066	INCALL I	2095553364	2:49	TESTCAMP	48
Sep/8-1	1091	ROCPA	8600062	DEAD		2:48	TESTCAMP	143
Sep/23-1	2123	ROCPA	8600069	INCALL I	4615552485	2:07	TESTCAMP	17
Sep/11-1	2027	ROCPA	8600071	INCALL A	9075554268	2:08	TESTCAMP	462
Sep/20-1	1921	ROCPA	8600065	INCALL A	3235553720	1:24	TESTCAMP	498
Sep/12-1	2035	ROCPA	8600052	INCALL A	3235555817	1:23	TESTCAMP	569
Sep/22-1	2234	ROCPA	8600056	INCALL A	3235557914	1:23	TESTCAMP	860
Sep/19-1	1768	ROCPA	8600058	INCALL I	7145558186	1:06	TESTCAMP	127
Sep/2-1	1080	ROCPA	8600074	INCALL I	9075559714	1:00	TESTCAMP	86
Sep/24-1	2342	ROCPA	8600070	INCALL A	3235554118	0:34	TESTCAMP	472
Sep/17-1	2246	ROCPA	8600059	INCALL I	7145550864	0:22	TESTCAMP	827
Sep/11-1	2124	ROCPA	8600059	INCALL A	4605555072	0:18	TESTCAMP	461
Sep/14-1	1194	ROCPA	8600060	INCALL I	9075554620	0:18	TESTCAMP	11
Sep/10-1	1731	ROCPA	8600051	PAUSED		7:16	TESTCAMP	28
Sep/7-1	1029	ROCPA	8600059	PAUSED		0:04	TESTCAMP	14

23 agents logged in on all servers
 System Load Average: 1.79

- Agent waiting for call
- Agent waiting for call > 1 minute
- Agent waiting for call > 5 minutes
- Agent on call > 10 seconds
- Agent on call > 1 minute
- Agent on call > 5 minutes
- Agent Paused > 10 seconds
- Agent Paused > 1 minute
- Agent Paused > 5 minutes
- Agent on a dead call

VICIDIAL web client - Mozilla Firefox

File Edit View Go Bookmarks Tools Help

http://10.10.10.16/sgc/vicidial.php#

Logged in as User: 6666 on Phone: SIP/138pcom to campaign: TESTCAMP LOGOUT

VICIDIAL SCRIPT 2006-01-12 16:02:45 session ID: 8600100 **LIVE CALL**

STATUS: Incoming: 7275554032 UID: VD112160143000726926

PAUSE RESUME seconds: 50 Channel: Zap/25-1 Cust Time: JAN 12 4:02:45 PM

RECORDING FILE: 60112160151_6666_7274614032 Customer Information:

RECORD ID: 896316 Title: Mr First: Matt MI: Last: lead01

STOP RECORDING

WEB FORM Address1: 1234 Fake 22

PARK CALL Address2: St Address3: 7275551214

TRANSFER - CONF City: 1234 West ~11@#%\$ State: CI PostCode:

HANGUP CUSTOMER Province: FL Vendor ID:

Phone: 7275551212 DialCode: 1 Alt. Phone: 7275551213

Show: test@test.com Email:

SEND DTMF Comments: comments go here

TRANSFER CONFERENCE FUNCTIONS:

INTERNAL CLOSER LOCAL CLOSER CODE HANGUP XFER LINE HANGUP BOTH LINES

NUMBER TO CALL: 7275551215 SECONDS: CHANNEL: DIAL OVERRIDE

DIAL WITH CUSTOMER PARK CUSTOMER DIAL LEAVE 3-WAY CALL BLIND TRANSFER VM

VICIDIAL web-client version: 1.0.59 BUILD: 51229-1028 Server: 10.10.11.11 HOT KEYS INACTIVE

LIVE CALLS IN YOUR SESSION:

#	REMOTE CHANNEL	HANGUP
1	SIP/138pcom-1fd3	HANGUP
2	Local/78600100@demo-17f0,2	HANGUP
3	Zap/25-1	HANGUP

Read 10.10.10.196

Setting Up, The easy way.



BUT...

If you are not a age old call center expert. you may lost in jargons.

There are lots of new thing to learn and customize to make this thing realy usefull for you.

Otherwise you will find yourself in a airplane cockpit for the first time in life and have to land this thing without a support.

Agenda



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Build it yourself.



To build your call center app. first try to know what you really want.

- Understand the need of you call center.
- Assume or calculate the number of calls you have to handle.
- Detarmine how much agents you require
- Fix working hour.
- Devide agents into groups based on your service criterion
- Understand the need of an IVR
- If outbound select the method and decide if you require predictive dialar.

Build it yourself.



After you decided all that primarily required. I assume you know how to install asterisk. and have basic idea about asterisk dialplan.

You may question why asterisk?

Its because its the most widely used open source telephony platform and its also the core of other well spread open source call center like vicidial.

Its simple to learn.

You may also used other open source telephony application like freeswitch, yete etc. all of them has its own call center module.

Build it yourself.



I think you already have your asterisk server running and giving you office PBX solution.

And now you want to add a ACD to it and distribute all your customer support calls to more than one extension orderly.

The very basic things you may need. is a Q.

In asterisk everything a Q will do is controlled by a configuration file called queues.conf

But before going to queues.conf will set our goal of what we are going through in this part of the presentation.

Build it yourself.



- Configure our first ACD.
- Add static agents to the Q
- Write a dial plan to send calls to that Q.
-
- Make the agents dynamic.
- Make them able to login and logout.
- Make them able to pause and unpaue.
- keep logs on database.
-
- Make calls recorded.
- Make managers able to SPY or Whisper on a call.

- Try to do some SBR (Skill Based Routing)
- CTI
- Predictive dialing basics.
- Call Progress Analysis Ideas.

Build it yourself.



Configuring first ACD.

```
[q_sample]
announce-frequency = 30
periodic-announce-frequency = 15
announce-holdtime = yes
announce-position = yes
announce-position-limit = 1
announce-round-seconds = 10
periodic-announce = queue-periodic-announce
queue-youarenext = queue-youarenext ; ("You are now first in line.")
queue-thereare = queue-thereare ; ("There are")
strategy = rrmemory
timeout = 20
retry = 5
maxlen = 0
ringinuse = no
announce-frequency = 0
announce-holdtime = no
servicelevel = 15
monitor-type = MixMonitor
monitor-format = wav
wrapuptime = 5
music = default
```

Build it yourself.



The most important part is strategy

ringall - ring all available channels until one answers (default)

leastrecent - ring interface which was least recently called by this queue

fewestcalls - ring the one with fewest completed calls from this queue

random - ring random interface

rrmemory - round robin with memory, remember where we left off last ring pass

linear - rings interfaces in the order specified in this configuration file.

If you use dynamic members, the members will be rung in the order in which they were added

wrandom - rings random interface, but uses the member's penalty as a weight when calculating their metric.

Build it yourself.



Add static Members/Agents

After the queue configuration add member like this.

```
member=> SIP/2000,Mr. ABC
```

```
member=> SIP/2001,Mr. EFG
```

```
member=> SIP/2002,Mr. HIJ
```

If you command `queue show q-sample` it will give you this output in asterisk console.

```
q-sample has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s talktime), W:0, C:0, A:0, SL:0.0% within 15s
```

```
Members:
```

```
SIP/2002 (Unavailable) has taken no calls yet
```

```
SIP/2000 (Unavailable) has taken no calls yet
```

```
SIP/2001 (Unavailable) has taken no calls yet
```

```
No Callers
```

Build it yourself.



Write Dialplan...

in your dialplan write this ...

```
exten => _YOUR_NUM,1, Answer()  
same => n,Queue(q-sample,tTwi)  
same => n, Hangup()
```

you are on.....

Make sure all three agents are in live sip extension. you will start to receive calls in an arranged manner.

Build it yourself.



Make Agents Dynamic.

To do so first you have to remove the members from the queues.conf. and then try something new in your dialplan.

```
login >>>>
```

```
exten => 11,1,AddQueueMember(q-sample,SIP/${CALLERID(num)})
```

```
logout >>>>
```

```
exten => 11,1,RemoveQueueMember(q-sample,SIP/${CALLERID(num)})
```

Build it yourself.



Pause >>>>

```
exten => 11,1,PauseQueueMember(q-sample,SIP/${CALLERID(num)})
```

UnPause >>>>

```
exten => 11,1,UnpauseQueueMember(q-sample,SIP/${CALLERID(num)})
```

Build it yourself.



To get reports of what happening there inside queue you may like to have logs.

To get the data on database you can use odbc or asterisk realtime. configure this accordingly. create database for queue

```
CREATE TABLE `queue_log` (  
  `id` int(10) unsigned NOT NULL auto_increment,  
  `time` char(10) unsigned default NULL,  
  `callid` varchar(32) NOT NULL default "",  
  `queuename` varchar(32) NOT NULL default "",  
  `agent` varchar(32) NOT NULL default "",  
  `event` varchar(32) NOT NULL default "",  
  `data` varchar(255) NOT NULL default "",  
  PRIMARY KEY (`id`)  
);
```

you will find records like this.

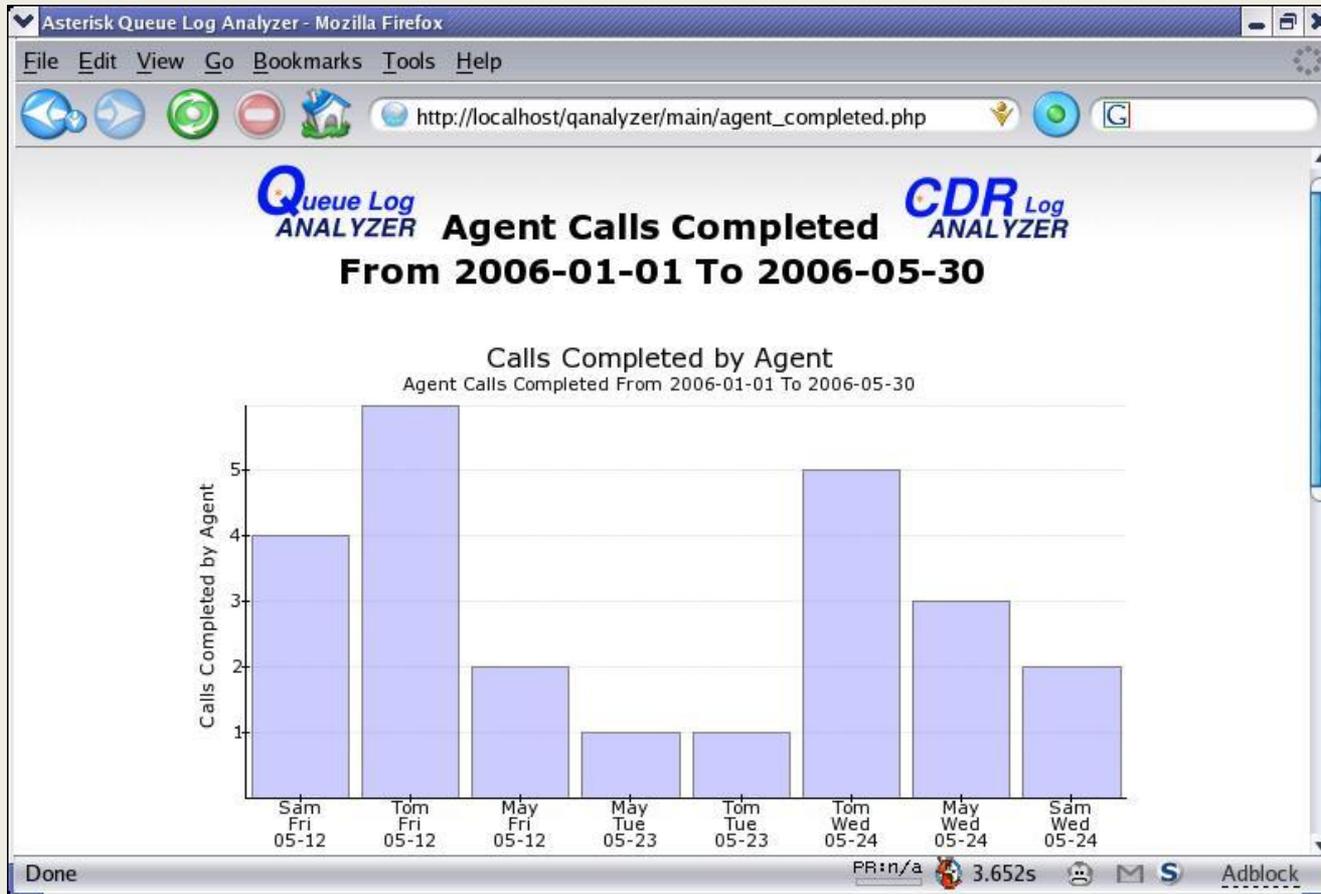
```
mysql> select * from queue_log;
```

```
+-----+-----+-----+-----+-----+-----+  
| id | time      | callid      | queuename      | agent | event      | data |  
+-----+-----+-----+-----+-----+-----+  
| 1 | 1198356717 | 1198356717.0 | q-sample | NONE | ENTERQUEUE | |serg |  
| 2 | 1198356719 | 1198356717.0 | q-sample | NONE | ABANDON     | |1|1|2 |  
+-----+-----+-----+-----+-----+-----+
```

Build it yourself.



You can write your front-end or use any open source one. There are some fine queue log analyzing software.



Build it yourself.



Recording Calls is important for call centers. to do so just add this lines before sending calls to the queue in your dialplan.

```
exten => _YOUR_NUM,1,Set(CALLFILENAME=${UNIQUEID})  
same => n,MixMonitor(/data/${CALLFILENAME}.wav,b)  
same => n,Queue(q-sample,tTwi)  
same => n, Hangup()
```

Build it yourself.



Spying agents helps manager to know how is your agent serving the customer. event he/she can help the agent server better inside a live call.

```
exten => _99XXXX,1,ChanSpy(SIP/${EXTEN:2},d)  
same => n,Hangup()
```

variations

- 4 -- Spy Mode
- 5 -- Whisper Mode
- 6 -- Barge Mode

Build it yourself.



Skill Based Routing.

You can route a call to specific agent/agents through SBR.

The easiest way to do that is to make some more queue based on skill set. and use an IVR to get customer feed back of what actually he want. so you can route him/her to specific queue.

The other idea are using database to query client information to get what kind of information he require and route him/her to specific queue or agent.

Agent penalty is a good way to route calls to specific agent.

Build it yourself.



Computer Telephony Integration is a complex thing. But for now you can do this thing in a simple way. Use IAX Clint insted of sip. Because IAX has a very fine app which is

SendURL (URL , option)

With SendURL you can send a url to the agent dialer. Most of the IAX soft phone will give you popup with a url to open. You can pass your CRM url with callers CID and other input he/she gave and with the help of IAX Softphone and a browser you can open the specific page for the client instantly.

Isn't it cool.

Build it yourself.



Here comes the tough part. Predictive dialer is a complete application that will dial for you and send only the answered calls to the agent in the queue.

For this you need some in-depth knowledge about AMI (Asterisk Manager Interface) and have to use Originate command to make calls.

You will need database knowledge and some very good application development skills too.

But for very basic call center who bothers deploying a predictive dialing option.

Build it yourself.



CPA is more complex than building Predictive Dialer. there is some open source one but I never find one that I can use. asterisk has some built in tool to do so.

- app_amd - Answering Machine Detection
- Waitforsilence - Silence Detection
- NVFaxDetect - Fax Detection
- NVMachineDetect - Answering Machine Detection
- NVBackgroundDetect - DTMF, Answering Machine and Fax Detection while playing audio file

But none of these are 100% accurate.

You can use some commercial one that are available. but try yourself before buy.

Any Question?



for more reading

Asterisk queues.conf

<http://www.voip-info.org/wiki/view/Asterisk+config+queues.conf>

Asterisk Wiki

<https://wiki.asterisk.org>

Thank You



Keep in touch with

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