



Reliable Technologies.

Date:

Ref No.

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Where quality is never compromised....

Reliable Technologies

102 7th Main, Brindavan Extn, Near Ganapati Temple, Mysore 570020

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Call Center Solution

VoIP has bridged the divide that existed between voice and data communication networks: effectively providing enormous opportunity in the quest to move towards creating synergistic combination of voice (and telephony features), data (and productivity applications) and video onto a single network. These previously separate technologies are now able to share resources and interact with each other creating new efficiencies.

This synergy has now been made more accessible - especially to end-users in small and medium enterprises, even individuals in emerging

markets - with the advent of strong and feature-rich open source technologies built on open standards.

At Reliable, we firmly believe in the power and ability of open source and open standards to deliver feature-rich, readily scalable, innovative and cost-effective solutions to technological challenges without compromising on quality. The efforts of its team of professionals with our cumulative experience culminated in a converged communication platform that delivers voice, data, instant messaging, and video built on a modular architecture that allows for scalability, integration of desktop and business applications and much more!

The Call Center Suite

With a powerful voice communication layer back-ended by a Customer Relationship Management module, the Reliable Call Center Suite (CCS) offers you an extremely cost-effective and user friendly way to stay in touch with your customers!

By employing a multi-tier architecture, the CCS allows you to choose precisely those modules that you require to get your call center kick-started and later on scale as your needs grow. The CCS cluster works equally well with VoIP and TDM circuits giving you a better value for money in terms of reusing resources wherever possible.

The very nature of the various components used in the CCS lends itself to setting up of solutions over a cluster of easily available and inexpensive computing hardware; thus not only reducing the total cost of investment but also ensuring high-availability with multiple levels of redundancy and load-balancing.

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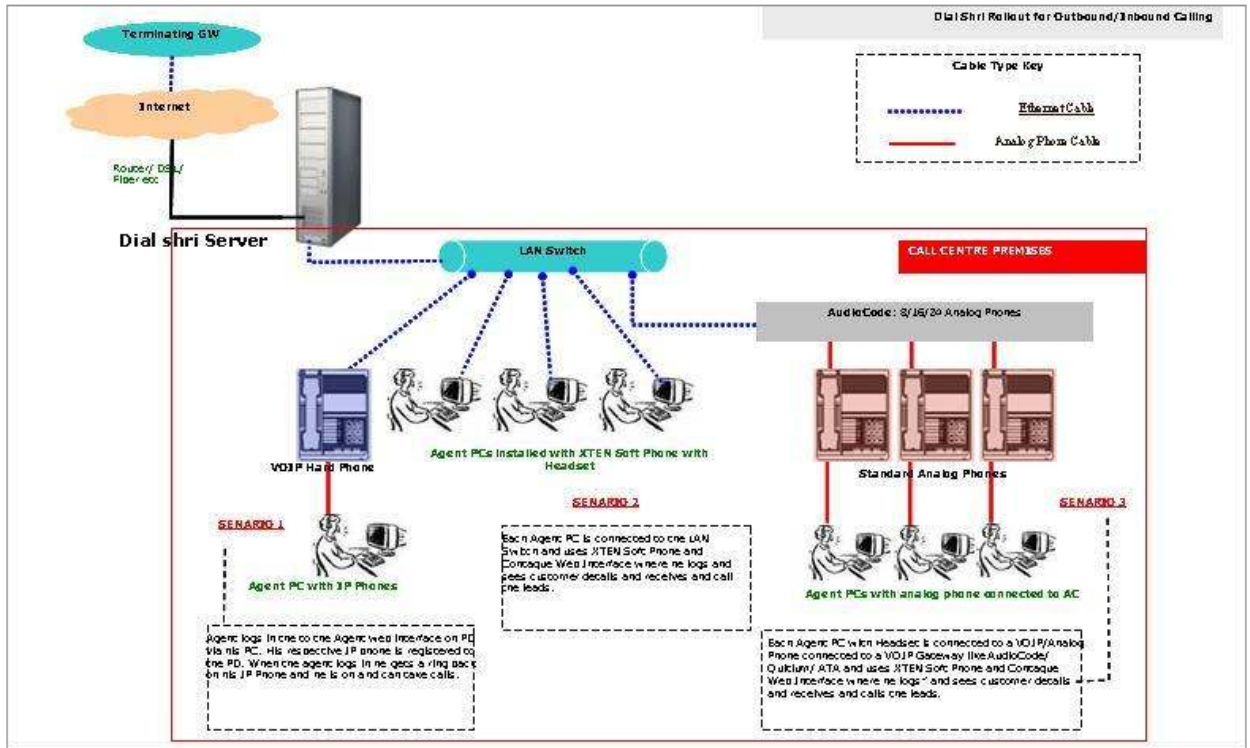
Sample setup (30 seat International Call Center):

The network diagram of an international blended (combination of outbound and inbound) call center is indicated below. Leveraging VoIP termination services delivered over standard leased lines saves point-to-point expensive IPLC charges.

The CCS cluster - accessible through a standards compliant browser and soft phone - is served by a separate database server (cluster) ensuring high availability and scalability.

The PBX engine of the CCS distributes the calls to the agents based on a customizable logic. The soft switch transports the calls over to the VoIP provider who will then provide interconnect to the TDM network

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Features and Specifications of Predictive dialer

- § Ability for an agent to call clients in succession from a database through a web-client
- § Ability to display a script for the agent to read with fields like name, address, etc. filled-in
- § Ability to set a campaign to auto-dial and send live calls to available agents
- § Ability to dial predictively in a campaign with an adaptive dialing algorithm
- § Ability to dial on a single campaign across multiple Asterisk servers, or multiple Campaigns on a single server
- § Ability to transfer calls with customer data to a closer on the local system or a Remote Asterisk server
- § Ability to open a custom web page with user data from the call per campaign
- § Ability to autodial campaigns to start with a simple IVR then direct to agent
- § Ability to park the customer with custom music per campaign
- § Ability to send a dropped call to a voicemail box per campaign if no agent is Available
- § Ability to set outbound CallerID per campaign
- § Ability to take inbound calls grabbing CallerID
- § Ability to function as an ACD for inbound and fronter/closer verification calls
- § Ability to have an agent take both inbound and outbound calls in one Session (blended)
- § Ability for agents to log in remotely and have calls redirected to any phone number
- § Ability to start and stop recording an agent's calls at any time
- § Ability to automatically record all calls
- § Ability to call upto two other customer numbers for the same lead
- § Ability to schedule a callback with a customer as either any-agent or agent-specific
- § Ability in Manual dial mode to preview leads before dialing
- § Ability for agents to be logged in remotely anywhere with just a phone and a web browser
- § Faster dispositioning of calls with agent key-binding (HotKeys)
- § Definable Agent Wrapup-time per campaign
- § Ability to add custom call dispositions per campaign
- § Ability to use custom database queries in campaign dialing
- § Recycling of Busy calls at a specified interval without resetting a list
- § Dialing with custom TimeZone restrictions including per state and per day-of-the-week
- § Dialing with Answering Machine Detection, also playing a message for AM calls
- § Multiple campaigns and lead-lists are possible
- § Option of a drop timer with safe-harbor message for FTC compliance
- § Variable Drop call percentage when dialing predictively for FTC compliance
- § Internal DNC list can optionally be activated per campaign
- § All calls are logged and statuses of calls are logged as well as agent time breakdowns



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- § Load Balancing across multiple inbound or outbound Asterisk servers is possible
- § Several real-time and summary reports available
- § Real-time campaign display screens
- § 3rd party conferencing (with DTMF macros and number presets)
- § 3rd party blind call transfer
- § 3rd party conferencing with agent drop-off
- § Ability to set user levels and permissions for certain features
- § Ability for managers to listen-in on agent conversations
- § Ability for managers to enter conversations with agents and customers
- § Web-based administration
- § Client web-application is available in English, Spanish, Greek, German, French, Italian,
- § Polish, Portuguese and Brazillian Portuguese
- § Admin web pages available in English, Spanish, Greek and German

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Features of PBX

ADSI On-Screen Menu System
Alarm Receiver
Authentication
(IVR)
Automated Attendant
Blacklists
Blind Transfer
Call Detail Records
Call Forward on Busy / No Answer
Call Forward Variable
(OSP)
Call Monitoring
Call Parking
Call Queuing
Call Recording
Call Retrieval
Call Routing (DID & ANI)
Call Snooping
Call Transfer
Call Waiting
Caller ID
Caller ID Blocking
Caller ID on Call Waiting
Calling Cards
Conference Bridging
Database Integration
Dial by Name
Direct Inward System Access
Distinctive Ring
Distributed Universal Number Discovery
for Message
(DUNDi)
Do Not Disturb
E911
ENUM

Fax Transmit and Receive
Flexible Extension Logic
Interactive Voice Response

Local and Remote Call Agents
Macros
Music On Hold / On Transfer
Predictive Dialer
Privacy
Open Settlement Protocol

Overhead Paging
Protocol Conversion
Remote Call Pickup
Remote Office Support
Roaming Extensions
Route by Caller ID
SMS Messaging
Spell / Say
Streaming Media Access
Supervised Transfer
Talk Detection
Text-to-Speech (via Festival)
Three-way Calling
Time and Date
Transcoding
Trunking
VoIP Gateways
Voicemail
Visual Indicator/Stutter Tone

Waiting
Voicemail to email
Voicemail Groups
Web Voicemail Interface



Voice transmission specifications

Computer-Telephony Integration	Protocols
AGI (Asterisk Gateway Interface)	IAX (Inter-Asterisk)
Exchange)	H.323
Graphical Call Manager	SIP (Session Initiation)
Outbound Call Spooling	MGCP (Media Gateway)
Protocol)	SCCP (Cisco® Skinny®)
Predictive Dialer	Traditional Telephony
Control Protocol	E&M
TCP/IP Management Interface	E&M Wink
Scalability	Feature Group D
Interoperability	FXS
- TDMoE (Time Division Multiplex	FXO
over Ethernet)	GR-303
- Allows direct connection of Asterisk	Loop start
PBX	Ground start
- Zero latency	Kewl start
- Uses commodity Ethernet hardware	MF and DTMF support
Voice-over IP	Robbed-bit Signaling (RBS)
- Integration of physically separate	
installations	
- Uses commonly deployed data	
connections	
Types	
- Allows unified dialplan across	PRI Protocols
multiple offices	4ESS
Codecs	BRI (ISDN4Linux)
ADPCM	DMS100
G.711 (A-Law & μ -Law)	Euro ISDN
G.723.1 (pass through)	Lucent 5E
G.726	National ISDN2
G.729	NFAS
GSM	
iLBC	
Linear	
LPC-10	
Speex	