SIP Based Call Centres – A vendor independent architecture for multimedia contact centres

Neill Wilkinson BSc, MSc
Quortex Consultants Ltd

The current call centre market is populated with vendor specific solutions, making the decision for the telecoms manager on functionality very difficult. Finding the best fit and being able to combine components from different vendors can quickly become a major systems integration project.
*This paper outlines the possibility for a vendor independent architecture that will allow best of breed components to be combined and allow for a close integration of systems without performing a major integration project. The architecture proposed also has the ability for rapid development of new applications for call centres by skilled Web developers [**not systems integrators or expensive software houses!**] and provide a framework for eCRM products.*

1 Introduction

Over the last decade Computer Telephony Integration (CTI) has rapidly become the way in which call centres have achieved the benefits of being able to combine telephony services with IT systems.

Whilst this growth in CTI has improved customer care and call centre sales capabilities, the industry has been marred by proprietary systems and non-standard interfaces, creating spiralling system integration costs.

In the background to this has been the rise of the Internet revolution and the emergence of IP as the way to connect systems.

The growth of the World Wide Web from Tim Berners Lee' original idea of a hyperlink information store to that of a powerful tool for application development, has allowed for dynamic content to be generated and constructed based on inbound context.

Now that Internet telephony is becoming mainstream, the major telecommunications operators are looking to replace their circuit switched infrastructures with packet telephony networks [1].

This is allowing CTI, the WWW and Internet telephony to be combined to create a new breed of call centre system. Presenting an opportunity to utilise the open nature of Internet telephony for call centres and to move away from proprietary protocols and vendor implementations.

The call centre market has seen significant growth over the last decade. Recognition of the need to handle customers efficiently has lead to the ideas of Customer Relationship Management (CRM).

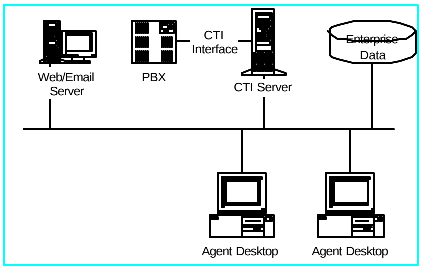
Whilst the idea of customer relationship is not that new, the linking of customer relationship to the realisation that retaining customers is more cost effective than trying to win new ones, is one that has captured the call centre industry.

The recent growth in the so-called e-economy has created a new area for customer contact, that of the web and email. Whilst this has created a new channel to market, it has also created a new need to ensure good customer care through this medium. This has led to the growth in the eCRM market.

> ***"The forecast for the eCRM market is estimated to be $6.1 billion at year-end 2000 and is believed to grow to an estimated $34.7 billion by 2005."*** **[2]**

2 Evolution of CTI architectures

The *de facto* architecture for Computer Telephony (CT) 3rd party call control is shown in the picture below.



This architecture involves a CT server communicating with a PBX or ACD system via a proprietary (vendor specific) interface, and a proprietary interface between a workstation application and the CTI server.

The back office enterprise database is accessed through, ODBC, JDBC or a direct SQL interface, either by workstation applications or by the CT Server. This enterprise data is fed from Marketing and other channels and areas such as OLAP and data mining are very important in extracting value from this data, however this is outside the scope of this paper.

This ends in a complex integration project of all the disparate systems through a plethora of interfaces and middleware, some of which is vendor specific. This is clearly not an environment for rapid application development.

The rise of the UnPBX

A little over 3 years ago, the use of NT as an OS for call centre applications emerged as an opportunity to move away from proprietary PBX/ACD systems. "*We are total believers in the concept of Windows NT based UnPBXs. We fully expect that over time, smaller phone systems (PBXs and Key Systems) will be replaced by UnPBXs.*"[1]

A number of vendors seized this opportunity[2] and started to produce hardware and software alternatives to the box based ACD/PBXs. The UnPBX was born [3].

The rise of the UnPBX was the birth of the movement away from circuit switched proprietary ACDs to a more open IP based solution. However 3 years ago (and to an extent now) proprietary call control protocols still reign supreme[3] (see section 4 for how this last bastion can be broken).

The current trends in open ACDs owes a lot to these UnPBX vendors. The telecoms world has moved on from here and a more open carrier sized upgrade is taking place.

3 Next Generation Networks

A lot of words have been printed about Voice over IP and IP this and IP that. The plain fact is that IP has become the *de facto* standard for multimedia communications. This means that ways have been sought to carry voice *reliably* over IP. This has lead to the emergence of what are known as the next generation networks. What this means for Call Centre technology is that there is a greater opportunity to capitalise on the changes in the core network.

The new PSTN

Conventional circuit switched (TDM) networks are slowly being replaced by packet switched networks carrying both voice and data. These networks are being created from new access techniques such as xDSL and HFC cable modems and core carrier network technologies such as MEGACO[4]. The mobile networks have not been left out and the UMTS standards are based on an ATM transport, with increased focus being placed on data services (GPRS and EDGE; and WAP and i-mode).

Whilst this creates the opportunity for more integrated applications, it will also force the telecoms operators to consider redesigning all of their conventional telecommunications services.

Since most (if not all) major telecoms carriers operate separate voice and data infrastructures, major consolidation will be required to efficiently move the voice network from TDM to the packet infrastructure of the data networks. In order to get real value (read revenue) from this infrastructure investment, without pricing the services out of everyone's reach, the telecoms operators will need to look at technologies such as VPNs for creating virtual segregation of customer traffic.

There is a major market opportunity for a new breed of telecoms operator to emerge, the ***IP-***

1 Bill Gates, Microsoft, Computer Telephony Magazine, June 1997.

2 Dialogic, Brooktrout, Netspeak et al.

3 Nortel – Compucall & Meridian Link, Aspect Application Bridge and Event Bridge, Lucent ASAI, Netspeak Communications Protocol (NSCP), et al.

Telco. This operator could emerge from an existing network operator, or could be launched from a global ISP like AOL. The IP-Telco could operate in a model similar to that of the MVNO[4] or switchless resale company.

To capitalise on the VNO model incumbent and competitive global telcoms operators could deliver telecommunications services on a wholesale basis from their networks, re-branded by VNOs completing the evolution of the switchless resale model.

The cost of entry for a new player to come into the market place as a global IP-Telco is relatively small and the potential for large revenues is huge[5]. The revenue is rapidly becoming derived from the services offered, not in the cost of access to the services[6].

This means a new entrant could enter the market place without having to own any network, they *just* have to negotiate peering access to a number of key POPs around the world and piggy-back their service on other incumbent networks. Clearly new peering arrangements that include QoS and service level specifications will have to replace the current ISP peering model.

Voice based information services & the new answering machine?

A number of the main vendors of IVR systems are looking to add packet voice interfaces to their products.

This is being augmented with XML based scripting in the form of VoiceXML[7]. This allows voice prompt scripts to be written in a platform independent language that also allows a close integration with Web servers.

Unified Communications is an application that has received a great deal of press recently. UC is Unified Messaging come of age. It provides the ability to integrate voicemail, e-mail, fax, SMS and paging into a single *inbox*. Work has been done on the use of SIP and RTSP for a UM platform [5].

UC looks set to replace company voice-mail systems and to provide a new answering machine service to users of in-net services via an ASP delivery model. BT offered an early version of an in-net answering machine with their CallMinder™ product.

A number of ISPs are starting to offer UM services for combining Fax and e-mail integration and allowing customers to dial in and listen to their email. The problem with these at the moment is that they require their users to have another access number.

4 The vendor independent CTI architecture

Building on the foundations of the UnPBX, the softACD has appeared and companies such as CosmoCom[8] are successfully utilising software frameworks such as Microsoft DCOM, J2EE and CORBA to develop conventional ACD functionality and combining Webserver capability to deliver eCRM.

These first generation softACDs are creating a new opportunity for vendor independent CTI capability, by combining the CT Server and ACD into a single platform build from open software frameworks.

The architecture of a truly vendor independent solution is available now, through products such as Sun Microsystems J2EE, Oracles' 8i or IBM's Websphere and BEA Weblogic.

What vendors will do is to write code on top of these application frameworks, providing the value-add for call centre applications. Application frame works such as this are also big business and companies such as BEA systems have made significant inroads into this market segment.

The open source movement

By combining open source H.323 and/or SIP stacks anyone with software engineering skill can develop a softACD and provide Web integration to deliver an eCRM solution. The biggest challenge is adding the value through reporting tools and remote maintenance capability and combining these with sound back office integration.

This is creating an opportunity for web developers to create new services, building on top of platform libraries without having to resort to

[4] Mobile Virtual Network Operator
[5] In economic theory, in a perfectly competitive environment, if excess profits exist and barriers to entry are low, competition will eventually appear and erode excess profits.
[6] Local call charges are rapidly being dropped for Internet access and broadband is being deployed on fixed fee packages.
[7] A.k.a VoxML
[8] http://www.cosmocom.com

help from systems integrators. This also means small businesses need not worry about large bills from system integrators and can take services from network based vendors for a fraction of the cost.

This also opens up the possibility for large telcoms operator to take the code developed by the new entrants to the market and transport it the large network based call servers, thus delivering large scale feature rich network based softACDs. This also creates the opportunity for ASP delivery of customer contact systems for the new e-economy.

It also opens up the way for systems/software houses to enter 3rd party development and partnership agreements with large operators to allow for the rapid delivery of new differentiated services.

Whilst the traditional role of the systems integrator is diminishing, their role is evolving from technology integration to business integration. Moving them up the value chain.

All this software is of course nothing without the call control protocol. SIP [6] is emerging as the favourite to fill this space. However H.323 is probably currently the most prevalent implemented call control protocol, for comprehensive comparisons of these protocols refer to [DALGIC].

Sun Microsystems are working on a JAVA based model called JAIN that incorporates these architectural ideas, including a SIP and H.323 stack for Intelligent Network based services.

The protocols responsible for converged communications are best viewed as a layered model [7]. This model consists of the call processing protocols, user protocols and support protocols.

Call Control Protocols

The call control protocols have been highlighted as SIP, H.323 and MEGACOP.

Of these SIP is arguably the most versatile and extensible as it is text based and designed around simplicity. This however does leave out a lot of embedded feature capability.

Why use H.323?

H.323 is actually a collection of Protocols and H.245 is actually the call control protocol (based on Q.931) combined with H.225. This framework has evolved from the work on the H.320 video conferencing standards, which means a lot of capabilities and features, are supported by the H.323 family of protocols.

Also H.323 is important because of its prevalent implementation by Microsoft both in it's Netmeeting™ product and in the TAPI 3.0 implementation on Windows 2000.

SIPs attractiveness for call control

One of the key issues of any Internet based service is security. SIP can support security through the use of any encryption standard. For example the widely available PGP package could be used to encrypt the signalling payload. One of the potential issues with SIP is that IP addresses are carried in the header message; this can be overcome with a SIP-NAT module (a version is available for Linux under the GNU licence). Solutions to this are far from elegant – yet [8], and it may be necessary in a network operator's environment to implement a firewall control protocol.

The SIP architecture has two key components that provide for load balancing and service transparency, namely: proxy servers and redirect servers for load balancing.

SIPs openness and extensibility allows for easy extension. The SIP INFO method [9] for in-call messaging is an example of this and is a perfect candidate for in-call communication between an agent desktop and a Proxy server SoftACD.

Support Protocols

The list of support protocols is endless, as this is essentially the list of all the specifications of the IETF. That said the protocols of most significance are: IP multicast for agent monitoring and 3-party call connect; The IPSec specifications for the creation of IP-VPNs and MPLS for the same purpose; The Real Time Streaming Protocol and Real Time Control Protocol for playback of recorded messages and finally the IRC client protocol for real-time chat between customers and agents.

One could bundle in the plethora of web protocols such as HTTP(S) however thess are more related to the web server delivery of content, Rather than collaborative interaction.

Application Frameworks

A word of caution on the title of this section, application frameworks are an area of study relating to the use of Object Orientated techniques for the production of domain specific software components. I do not intend to expand on this here, as this is a massive topic area in its own right. If application frameworks interest you I suggest you refer to [MOHAM].

This section highlights some of the middleware frameworks that are available from development of the collaborative applications necessary to support an eCRM service.

Firstly Sun's J2EE coupled with Sun's iPlanet Web server. These products build on Sun's Java 2 virtual machine for the rapid development and deployment of component based applications and creates and environment for the creation of an integrated ACD call server.

The BEA/Nokia and Motorola (MIX) platforms use the same capabilities of Java Server Pages and Java Sevlets to create an integrated environment for Wireless platform application development. These could be used as the basis of Wireless communications server.

Finally DynamicSoft's Proxy Server products utilise CPL and SIP CGI for call control in a SIP environment. It is a small mental step to enhance this to a Java Serverlet environment to provide softACD functionality on top of this platform and to link these platforms to web servers to provide multimedia integration. Queue and agent skill scripting could be provide by CPL[10,11].

Call Servers

The evolution of the PSTN from a collection of circuit switches to a distributed call server (Media Gateway Controller) and gateway (Media Gateways and Signalling Gateways) architecture; and the use of Proxy Servers in the SIP architecture. Has created the opportunity to deliver all services from Call Servers.

This might be considered as an issue with "all your eggs in one basket", but that is not the case because the call servers are in their own right fault tolerant distributed platforms.

What is key is that the call servers are being constructed from *standard* IT hardware. This opens up the possibility for these servers to be used for network-based softACDs.

Billing for Network Services

Clearly from a network operator's perspective, being able to charge for the use of a service is very important. Billing for next generation network services will be performed by the call servers (MGCs, Softswitches, SIP-Proxy Servers, H.323 gatekeepers). The difficult arises from the fact that once an end-point has discovered the other parties end-point there is theoretically nothing stopping the originating end-point by passing the call server, thus creating the opportunity for fraudulent use of network resources.

In order to prevent this in for example a SIP situation (a similar solution would work for H.323 end-points), then the SIP user agent must contact the SIP-Proxy via a NAT firewall, thus preventing any visibility of the "internal" network [12]. This of course leads back to the issues involving the use of firewalls, highlighted above.

5 Conclusions

Firstly let's start with areas not covered by this paper: Business Models (Marketing/Sales/Service Sale Cycle, when things go wrong, order tracking, Workflow); Delivery Models (ASP, CPE). These areas are papers in themselves and the author suggests the reader looks elsewhere for discussions of these topics.

Telecoms operators need to be looking at platform vendors rather than mega-software vendors.

The mega-software vendors will develop functionality based on what the mass market wants and customisation and differentiation will be more difficult than with a platform solution.

The platform solution has the advantage of allowing for vertical market products on a common architecture. The scale of product offering required by a telecoms operator makes the platform solution the obvious choice. For wholesale delivery of services to different customers, telecoms operators will also need to consider the ability to segment the services in a secure fashion.

Watch out for consolidation in this market space. The market is still young and vendors such as Cisco, Alcatel (Genesys products), Siebel and Clarify (Nortel Networks Company) have the power to swallow new entrants. The author has specifically steered clear of the CRM products provided by companies such as Siebel and Clarify, purely because the space these vendors occupy is currently the area of product and System integration, this white papers whole premise was to discuss the framework that will underlie these vendors products.

Also watch out for the platform people (BEA, Vignette and Art Technology Group), and for the architecture vendors, IBM (Websphere), Sun Microsystems (J2EE and I-Planet). The other big boys keen to push this opportunity are of course Oracle and Sybase. Oracle's 8i & 9i products could form the core of many eCRM offerings.

Another company in this space, that will later emerge, as eCRM candidate is DynamicSoft. DynamicSoft are the leaders in SIP based products that will under pin the voice side of softACDs (their Chief Scientist Jonathan Rosenberg is very active in this area and has recently proposed application architecture this may go someway to achieving this [13]). All of the other vendors mentioned are firmly from the IT and data management camp. So far these *IT* vendors have not integrated the voice portion of customer interaction, arguably the most important channel.

6 References

1. Financial Times Oct 3rd 2000: Telecoms operator aims for switch to Internet technology within 3 years.
2. UBS Warberg Global Equity Research - eCRM: Stop the insanity.
3. Edwin Marguiles – The UnPBX, Flatiron publishing, INC. ISBN: 1-57820-014-8.
4. RFC2885 – MeGaCo Protocol.
5. Unified Messaging using SIP and RTSP Kundan Singh & Henning Schulzinne, Columbia University.
6. draft-ietf-sip2543bis-02, Rosenberg et al. latest draft of RFC2543 – expires Jan 2001
7. Uyless Black – Internet Telephony (Call Processing Protocols), Prentice Hall, ISBN: 0-13-0225565-3.
8. draft-rosenberg-sip-enfw-00.txt, SIP traversal through residential & enterprise NATs and Firewalls. J Rosenberg, H Schulzrinne.
9. RFC 2976 SIP INFO method, Steve Donovan, Dynamicsoft.
10. Programming Internet Telephony Services, Jonathan Rosenberg, Jonathan Lennox and Henning Schulzinne. Tech Report Number CUCS-010-99.
11. draft-lennox-sip-cgi-01.txt, J. Lennox, J. Rosenberg, H. Schulzrinne, Columbia University/Bell Labs.
12. IP Telephony Packet-based multimedia communications systems. Olivier Hersent et al. Addison Wesley.
13. draft-rosenberg-sip-app-components-00.txt, An Application Server Component Architecture for SIP, Rosenberg, Mataga, Schulzrinne.

[DALGIC]
Comparison of H.323 and SIP for IP Telephony Signaling, Ismail Dalgic, Hanlin Fang, 3Com Corporation, Technology Development Center, 5400 Bayfront Plaza, M/S 3219, Santa Clara, CA 95052, Cisco Systems170 West Tasman Dr., San Jose, CA 95134

[MOHAM]
Building Application Frameworks – Object Oriented Foundations of Framework Design, Mohamed E.Fayad, Douglas C.Schmidt and Ralph E.Johnson; Publisher Jon Wiley & Sons – ISBN 0-471-24875-4

