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Unified Telecommunication Model ©
White Paper

Fourth in a Series on
The Major Components of Voice and Data Convergence in the Enterprise

Communication Portals

Multi-modal / Trans-modal Access and Management, including Presence

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Abstract:

This generic paper is the fourth in a series on migrating toward a ***Unified Telecommunication Model (UTM©)*** in the enterprise. The first paper was on *UTM©* for Communication Servers & Contact Distribution Systems; the second paper was on *UTM©* for VoIP Network Design, including WiFi & Mobility; and the third on *UTM©* for Self-service & Automated Response Systems – Legacy vs IP-based. Earlier papers serve as prerequisites to the topic at hand. Please contact CollabGen for copies of these or other white papers in the series.

Each part in the series will categorize a major component or set of components as an integral part of an overall technology architecture (infrastructure, systems and applications) and will cover the ***major trends in telecommunications including networks convergence, TDM / IP Telephony, portals, presence / proximity, multi-modal and collaborative communications.*** The central thesis is to look at traditional areas such as Telephony and Media Processing Systems (voice, fax, eMail, text, video etc.) and provide a holistic view into the current state-of-the-art for integrated / unified telecommunications. On the practical side, for most people it all comes down to the user interfaces and how well these systems work for them. For enterprise executives and staff, the goal of this series is to provide a better understanding of which components may need to be addressed and in what order.

This series will also cover issues associated with this new model including the efficacy of implementing various components in-house or via a service provider, and mitigating concerns including network and systems security, quality and reliability.

Each White Paper is generally structured as follows:

I. Executive Overview

- Scope
- Benefits
- Processes
- Technology Issues

II. Analysis

- Encapsulates the component's technological domain
    - Primary enabling technologies
    - Standards
    - Applications
    - Configurations
- Next Steps

III. Optional section

- Specific vendor(s) solution set

The above sections contain perspectives for **enterprise, contact centers, and remote / mobile applications.**

Note to Readers

- **This symbol** (the CollabGen logo) designates **analysis checkpoints** SM that may be particularly significant to your organization's effectiveness and may warrant further professional / consulting services discussion.

Terms and acronyms identified in *italics* can be found in the attached Glossary.

Customized versions of this White Paper or Series are available upon request based on your specific environment and industry. Please contact CollabGen for other White Papers in this series.

Other reports include detailed side-by-side comparisons of telecom and contact center vendor solutions based on CollabGen's proprietary analysis criteria.

About CollabGen

CollabGen Inc. (www.collabgen.com) performs research and analysis of telecommunication technology, systems and applications. We provide management consulting services for clients to improve communications and customer service in contact centers and across the enterprise regardless of size or complexity. In addition to knowledge transfer and independent consulting, we are certified to provide Voice Response and Web Usability Audits and re-design services based on video-based testing to establish Best Practices. We are also certified in Benchmarking for Contact Centers to determine efficiency and effectiveness in comparison with your peer organizations and provide recommendations for improvement. CollabGen is an alliance-based firm comprised of independent analysts and consultants. Independent telecom planning, procurement assistance and project management is provided in conjunction with its subsidiary, eTelecom Consultants (www.4etel.com).

About the Authors

Ed LaBanca – CollabGen

Ed is President and Principal Analyst / Consultant for CollabGen. He has over 30 years experience working for both large and small equipment manufacturers and software development organizations in the computing and telecommunications industries including 6 years as an independent analyst / consultant. His current emphasis is on voice / data convergence technologies and products related to contact centers and areas where the Internet, Public Switched Telephone Network, front office and back office systems merge. His focus is on Interactive Voice Response (IVR) / Interactive Web Response (IWR), Computer-Telephony Integration (CTI), Voice over Internet Protocol (VoIP) and Voice / Fax / Unified Messaging.

Arthur M. Rosenberg – The Unified-View

Art Rosenberg has been involved with pioneering interactive computer and telecommunications technologies since starting his computer career with the RAND Corporation. After helping to develop one of the first online time-sharing systems sponsored by ARPA, he became particularly focused on computer-based telecommunications technology for both personal call and message management services and call center services when he joined the market / product planning team for Delphi Communications Corporation, one of the industry's earliest technology developers for Voice Mail, multimedia messaging, and call center technologies in the late 1970's.

Art has been a well-known independent writer, speaker, analyst, and consultant in this subject area since 1983, when he helped develop the industry's first Voice Mail seminar for *Business Communications Review*. He has been a frequent contributor to *BCR, Gartner Group / Datapro*, and other industry publications since that time.

As a knowledgeable industry expert in the transition of traditional telephone call centers into multi-modal contact centers for e-commerce, Art also co-authored a best-selling book with well-known industry writer, Paul Anderson, on the practical role of customer communication technologies and Customer Relationship Management (CRM) concepts, “The Executive's Guide to Customer Relationship Management.”

Mark Levinson – VoxMedia Consulting

Mark Levinson is President of VoxMedia Consulting. He has over fifteen years software and telecommunications industry experience, including stints at Bell Laboratories and GTE, where he did research, software development, system design, and project management. During five years as Director of Systems Development and Director of Partner Support at SpeechWorks International (since acquired by ScanSoft), he managed and contributed to many groundbreaking projects covering almost every aspect of voice technology. He holds S.B. and Sc.D. degrees from MIT, and an MBA from Boston University.

November 2005

I. Executive Overview

Scope

With the convergence of Business Information Systems, the Internet / World-wide Web, and wired / wireless *IP Telephony* communications *(VoIP)* with text messaging, person-to-person telecommunications is being transformed in totally new ways in the context of a *Unified Telecommunication Model (UTM©).* As business people become more accessible with mobile devices, they are also becoming less available for real-time contacts, making *multi-modal* messaging alternatives more practical for both person-to-person contacts and information exchange. Making efficient contact with people, as opposed to accessing business applications (online information retrieval / transactions) is a significantly greater challenge for enterprise technologies. This paper addresses ***UTM© for Communication Portals***.

Computer-based call processing and messaging systems (primarily asynchronous Voice Mail and eMail) have been implemented extensively since the early 1980's on first generation, stand-alone systems in conjunction with *PBX's and* eMail servers within an enterprise (as discussed in the previous paper on *UTM© for Self-service & Automated Response Systems – Legacy vs IP-based*). However, such systems provided only limited capabilities to "contact initiators" who had to guess about the availability and *modality* of contact with a "contact recipient."

Communication Portals, which include telephone and online access to web-based messaging and real-time voice / video telecommunication applications, have just recently entered the market in the past couple of years. These applications are focused on making contact with people, as opposed to simple business information retrieval or self-service transactions, and provide facilities for both contact initiators and contact recipients to dynamically and efficiently manage accessibility by both people and automated business services. Automated self-service functions can also be incorporated within telephony (speech) and online web *Communication Portals* can be provided by both the enterprise organization and "public" service providers and communication carriers. They can be "federated" across enterprise and public networks to support the concepts of "virtual" Unified and *Trans-modal Communications*, which have been advanced for many years by one of this paper's contributors, Art Rosenberg.

Communication Portals provide telephone and online web interfaces for personalized, *presence*-based management of contact access availability, *modality*, and permission-based communications with individuals inside or outside an organization. They act as virtual, online "intelligent address books" that integrate personal calendars, personalized access "rules", and dynamically changing user status information. They support the convergence of a variety of communication device interfaces for integrating the use of *IP Telephony* voice and video conferencing with all forms of messaging. This requires interoperability between distributed services and enterprise systems in a "federated" operational environment across public and private networks. So, whereas Automated Attendant and Voice Mail were localized, enterprise solutions to answering / directing a voice call and relaying stored information, *Communication Portals* are designed to make all forms of messaging interoperate with conversational voice communications based on a user's *modality*, which as alluded to earlier, can be referred to as *Trans-modal Communications*. Furthermore, current speech recognition technology allows enterprise-wide Automated Attendants to offer voice access to thousands of names without any need to remember extension numbers, and makes possible other powerful speech- controls and navigation features. This paper also provides an introduction to *Unified Messaging,* along with current speech-enabled messaging and *Voice Portals* by Mark Levinson.

The advent of *Voice over Internet Protocol (VoIP)* and *IP Telephony* is enabling traditional proprietary telephony technologies to become "open," more software dependent, and more application-flexible. In addition to enabling traditional telephony functions to become "virtual" and centralized, which presents significant cost-saving opportunities to both enterprise organizations and service providers, *IP Telephony* and *IP* data infrastructures enable voice communications to converge at all levels of telecommunications:

- **Level 1** – **Networks** (public and private; enabling technologies for voice and data convergence among devices);
- **Level 2** – **Applications** (enterprise business application servers for information access / online transactions or person-to-person telecommunication application servers); and, last but not least
- **Level 3** – **The end user communication device / user-interface / media** (including software clients).

Level 3 relates to the ***modality*** of communications, primarily the user interfaces, applications and media employed with different devices, wired or wireless, either for person-to-person communications or for information exchange using enterprise business applications. These can be combined for group Conferencing and Collaboration (which will be the topic of a subsequent paper in the series).

> "An example of ***trans-modal*** communications is exemplified by the traditional "telephone answering" function of legacy Voice Mail systems that convert a real-time call attempt into an asynchronous voice message. Conversely, Voice Mail systems enable a voice message recipient to respond with a "call return" attempt to an outside caller who may or may not be an internal Voice Mail system user… Currently, communication address links are being embedded in text / *Instant Messaging (IM)*… for click-to-talk."

An example of ***multi-modal*** person-to-person communications is eMail. An eMail communication could be sent from a full-featured PC-based application and read from a mobile Personal Digital Assistant (PDA); or vice-versa. Alternately, it could be read from a basic cell phone using a Voice User Interface (VUI) and *text-to-speech (TTS)*.

An example of ***trans-modal*** communications is exemplified by the traditional "telephone answering" function of legacy Voice Mail systems that convert a real-time call attempt into an asynchronous voice message. Conversely, Voice Mail systems enable a voice message recipient to respond with a "call return" attempt to an outside caller who may or may not be an internal Voice Mail system user (provided the caller's number is identified). Currently, communication address links are being embedded in text / *Instant Messaging (IM)*. With convergence between eMail, *instant messaging*, voice messaging, and *IP Telephony*, this transitioning between communication modalities can be efficiently extended in real-time between the communicating parties, e.g., "click-to-chat" and "click-to-talk."

The combination of all three levels described above will serve to maximize telecommunication flexibilities and efficiencies, as well as improving intra / inter-enterprise collaboration.

More detail on *Communication Portals* and their correlation with *modality* vis-à-vis device, user interface, applications and media will be addressed in the Analysis section to follow.

Benefits

There are benefits for consolidating, combining and migrating from stand-alone application servers to *Media and Application Processing Servers* in conjunction with *Communication Portals*.

These include:
- Benefits to the enterprise
- Benefits to end users

For successful adoption of such new technology, both must realize value from the changes in technology.

Cost Reduction Benefits to the Enterprise

The enterprise is responsible for administration and support of internally used technology. For distributed organizations with multiple locations, legacy telecommunications technology has always been localized hardware, requiring local support. With *IP* network infrastructure, software functionality can be centralized and / or remotely maintained, thus reducing the problems and costs for supporting all forms of communications activity.

> "…wireless communications mobility is not a legacy enterprise technology and end users requiring such accessibility will not be waiting for any life cycles to end."

Legacy *TDM* technology still works effectively and most organizations are planning to phase out old technology in a "graceful" migration to *IP*-based infrastructure and applications. In other words, as life cycles for old technologies expire they will be replaced with *IP*-based technology. However, wireless communications mobility is not a legacy enterprise technology, and end users requiring such accessibility will not be waiting for any life cycles to end. For this reason, wireless carrier services and new, wireless handheld devices are appearing in greater numbers within enterprise organizations to supplement or replace traditional wired desktop communications.

Fully integrated, *multi-modal* communications will have to include wide area wireless services to support end user mobility in various ways, and that will require communication applications within the enterprise to interoperate seamlessly with those provided by carrier services, in the same way long distance services work with enterprise telephone systems.

Communication Productivity Benefits

Although not a "hard benefit," like technology procurement and support cost reduction, operational productivity that results from more efficient communications between people is becoming a more important facet of converged communications and unified telecommunications. With the scope of user productivity in terms of task performance, there are several levels of improvement for productivity.

Individual Productivity or "Micro-productivity"

This has always been an objective for simplifying the user procedures involved in how people make contact and exchange information. The main limitation of the past has been the dependency of communication upon location, particularly for voice telephony. Messaging, both voice and eMail, introduced the concept of the "virtual" mailbox, which can be accessed from anywhere. However, there are still limitations imposed by the type of device needed to gain access, and enterprise Voice Mail systems were not interoperable across systems or across enterprises as is the case with eMail.

Another major source of user communications efficiency is in embedding all the necessary information users require to initiate contact with another person within the framework of the communication device being used. This includes immediate access to contact information and entering such information into the network ("click to contact") as well as providing personalized context information for efficiently carrying out the communication transaction with a specific individual (contact history logs, access to files and documents associated with past contacts, etc.).

One form of this is the *Automated Speech Recognition (ASR)* enabled Automated Attendant. It allows voice callers to connect to the desired party (or the party's virtual *presence*) by just saying his or her name, rather than having to look up and dial telephone numbers and extension numbers. In our experience, even for internal calls in organizations of fewer than one hundred members, people often find it easier and faster to use the Automated Attendant than to dial extensions.

"…savings from converged, unified messaging capabilities could range from a half-hour to an hour a day. However, the real payoff to the enterprise will come from getting operational decisions and actions done more quickly to capture business opportunities."

One of the Micro-productivity benefits of wireless messaging is that it enables the productive use of "dead time" to catch up and respond to messaging contacts. Such dead time opportunities occur while traveling, waiting for events, meetings, etc.

Cross-modal messaging access, i.e., being able to retrieve, forward, or reply to text messages with a speech interface, further improved user time productivity opportunities for dead time. This was particularly useful for mobile users retrieving and replying to eMail text messages by voice while driving a car.

Depending upon user job activities, individual productivity savings from converged, *unified messaging* capabilities could range from a half-hour to an hour a day. However, the real payoff to the enterprise will come from getting operational decisions and actions done more quickly to capture business opportunities.

Group Productivity or "Macro-Productivity"

A more significant ROI for the enterprise can be realized when a time-sensitive group task can be completed more quickly even though members of the group are not co-located. Meeting deadlines can be critical and can result in generating revenues, avoiding penalties, or losing potential revenues. Unfortunately management metrics and tools for quantifying how efficiently groups communicate for such situations are still evolving. Since there are now several ways that people can communicate, such metrics must encompass all modalities of contact in order to assess the performance of operational activities. This is one of the potential areas that will benefit from a unified telecommunications environment.

Macro-productivity considerations drive the need for individual users to be more contact accessible and responsive to asynchronous messaging, since they can impact time-sensitive action by the group as a whole. This holistic view of personal accessibility management is particularly important for enterprise business communications, because it is not a matter of just individual user preferences, but what is needed to make the group more efficient in getting a decision made or a problem resolved in a timely manner.

With growing use of *multi-modal* wireless devices, there will be an increased need to be sensitive to individual user circumstances that may dynamically determine the *modality* of contact and response. Mobility preempts the notion that users can completely "pre-program" in advance how they can be contacted with fixed rules and calendar entries. Subject matter and contact originator priorities are a relative dynamic that will constantly be changing, and must therefore be subject to flexible, real-time user management.

Customer Contact ROI

> " 'click-to-chat,' and 'click-to-talk' options for live assistance… will be further facilitated as consumers increase their adoption of VoIP / Internet Telephony services."

A key area of interest for any enterprise telecommunications application is anything that enhances customer interactions with the company, since customers are where revenue generation originates, as opposed to just cost savings. As web eCommerce becomes a global 24x7 venue for online product and service information, as well as business transactions, telephone access by customers is giving way to web self-services, as well as "click-to-chat," and "click-to-talk" options for live assistance. The latter two options will be further facilitated as consumers increase their adoption of *VoIP* / Internet telephony services, making *"trans-modal"* access to customer assistance more flexible and efficient than traditional *TDM* telephony.

> "Using *ASR*, a few simple menus can be used to determine the proper routing of each call, whether to an individual, an appropriate customer service agent group, or *ASR*-based self-service applications."

Similar efficiencies and service improvements are now attainable for conventional telephone contacts with A*utomated Speech Recognition (ASR)* and network call routing. Multiple toll-free numbers can be consolidated into a single point of contact for all customer telephone interactions with an organization. Using *ASR*, a few simple menus can be used to more accurately determine the proper routing of each call, whether to an individual, an appropriate customer service agent group, or *ASR*-based self-service applications.

Furthermore, all voice-accessible company and customer-service information can be consolidated into *Voice Portals*, comparable to *Web Portals*, which can be easily navigated with *ASR*. In addition to call direction functions, *Voice Portals* can provide employee self-service transactions such as Human Resources and benefits; and customer self-service transactions such as account inquiries, service and product information, order entry, order status, etc.

Even relatively complex self-service transactions, like *knowledge-base* searches and technical support, are now becoming more practical with *ASR*-based systems. *Knowledge-base* queries inherently involve more free-form language than more structured interactions like account status or travel reservations. These interactions generally require that the system recognize and extract meaning from natural language phrases and sentences. This can be accomplished with *Statistical Language Models* and *Natural Language Processing*. Although these technologies are powerful, they are also subtle, and a good deal of skill and experience is necessary to deploy them successfully.

The most powerful implementations of *Voice Portals* are those where they are integrated with the other customer contact modalities. Navigation design and terminology is consistent between voice and the web. Calls not completed within the portal are transferred to customer service representatives along with data collected during the call to identify the caller and provide (via back-end data queries) all related customer information. In this way, the transaction is seamlessly acquired by the representative without frustrated customers being required to repeat the same information a second time.

These call-direction and self-service applications are now implemented with *Voice XML* and related web-based standards. Commonality with web applications promotes reuse of network and platform infrastructure, business logic software components, back-end interfaces, and development and maintenance tools, leading to radical system simplification and cost reduction relative to earlier, proprietary voice-response and telephony systems.

IP Telephony for customer contact will enable the enterprise-wide “virtual” contact center for both distributed customer-facing enterprise staff and outsourced staffing resources. In addition, assistance options for online customers can also exploit all forms of messaging, whether text messages or voice messages, when waiting for real-time connections is not necessary. Finally, business application services can exploit *multi-modal* information delivery to mobile customers through the power of *IP*-based *presence* and *modality* management.

IP Telephony Architecture ROI

- With a centralized *IP Telephony* architecture, applications can go beyond traditional applications with limited telephone sets to smart software clients that can identify an incoming call and allow real-time point-and click control for routing a call to an off-premise phone, cell phone etc., or a pre-defined “*CTI*-type” process.
- Applications are implemented or easily changed and extended using browser technology including drop-down menus, radio buttons, check boxes and wizards
- Increased built-in flexibility for continuity of business / disaster recovery, including outsourcing options
- Centralized application servers support consistency across a distributed network , as well as easy of management control
- Comprehensive end-to-end real-time and historical reporting and analytics
- Administrative and supervisory functions via web browsers
- Web multi-media services and standards, including those which enable toll-quality voice for the enterprise, provide richer and broader set of added capabilities for new and enhanced converged applications.

Current generation *Media and Application Processors / Servers* provide *IP* streaming, forwarding and integration of voice, data and video in various combinations in conjunction with *Web Services*. Therefore traditional *voice processing* applications such as Auto-attendant / Voice Mail; Fax; *IVR*; etc. can now be implemented and managed using browsers and thin-clients as well a through a Voice User Interface (VUI) under a single (unified) environment for all media, which naturally extends to *Communication Portals*.

Processes and the Addition of Personal / Group Communications Portal

The following is an overview of enterprise systems and applications, extending legacy *PBX* or *Communication Servers* and web site applications to a ***Communications Portal*** for personal and group communications management with access to business applications. Other interrelated systems include *Enterprise Application Integration (EAI), Customer Relationship Management (CRM), Unified Contact Distribution (UCD©) and Computer-telephony Integration (CTI).* These apply to contact centers and intra / inter-enterprise communications, including mobile workers:

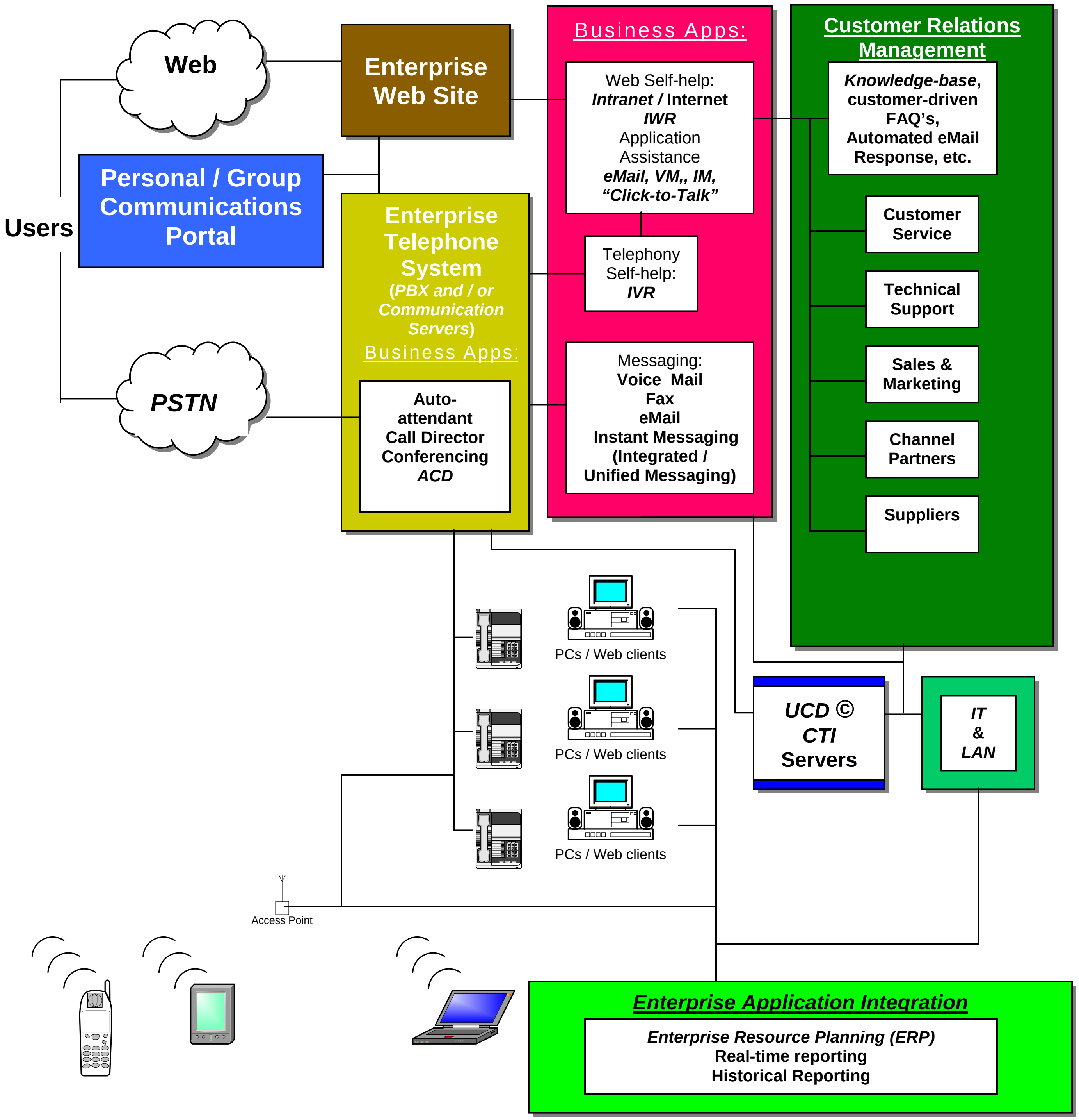


Figure 1

Applications can be incorporated within *Communication Servers* in conjunction with Web Servers, which can be centralized and distributed based on business requirements and the new opportunities being presented with networks convergence.

Together these servers can more efficiently accommodate all forms of multi-channel / multi-media communications (refer to the following Analysis / Configuration section for more detail).

Customer Relationship Management

The diagram above (Figure 1) also shows the correlation of a *Communication Portal* with business applications and functions that can be extended to customer self-service and knowledge transfer (both automated and semi-automated).

Person-to-person Communications using Presence

Business communication contacts between enterprise personnel who are geographically distributed have expanded in both real-time and non-real-time domains. Real-time communication now includes both wired and wireless text messaging [*instant messaging*, wireless eMail, *Short Message Service (SMS)*], while text messaging can be retrieved through speech interfaces and responded to with a voice connection or a voice message. Business applications that need to deliver time-sensitive information to a user can also exploit the flexibility of *multi-modal* contacts and *presence* in the same way that another user can, by using the power of a *presence* management server to deliver information on the most appropriate *modality* available.

For contact center applications, where live assistance or expertise is required, communications accessibility to any qualified and available enterprise contact, wherever they may be, can be exploited through the use of *presence* and *modality* management. This will be applicable to both intra-organizational needs, as well as for customer-facing requirements.

Technology and Application Issues

The principal technologies and applications used in *Communication Portals* are described in the following Analysis section. The primary issue is to align the enterprise business model to the *Unified Telecommunication Model*©, including people and process for your particular environment and to implement more effective communications and competitive differentiators at the network, application, and end-user device layers .

II. Analysis

Technological Domain

Communication Portals (web and phone-based) are categories within *communication and media processing* systems and applications. These systems enable the use of ubiquitous tools including various forms of telephones, notification devices and web clients for real-time, streaming and store-and-forward communications and collaboration. They provide individuals and groups a view into the *presence* and *modality* of their colleagues. Each user manages how and when they can be reached, including customized contact rules.

Primary Enabling Technologies

- Media (voice / sound, image / fax, and video) digitization and compression
- Touch-tone recognition and generation
- Voice and video streaming; and store-and-forward / retrieval
- *Automated Speech Recognition (ASR)* – Speech Recognition interprets callers' spoken phrases and matches them to sequences of words allowed by a grammar.
- Speech Synthesis / *Text-to-Speech (TTS)* – Speech synthesis (the reverse of speech recognition), is where computer-generated audio is created from text. The best current products sound fairly natural (compared with earlier, robotic-sounding voices), but still lack the intonation and subtlety of human speech. *Text-to-speech* is used primarily where it is impractical to pre-record the information, either because the number of possible variations is too large (like names and addresses) or changes too frequently (like weather reports). However, it is widely used in *Unified Messaging* for reading eMail via the phone.
- Packetized Voice and Data
- *Web Services*

Standards

There are various standards and *protocols* used for the *media* and *call processing* technologies itemized above which need to be considered in detail when specifying associated systems and applications to ensure adaptability and inter-operability (reference the first paper on *UTM© for Communication Servers & Contact Distribution Systems*).

For *Communication Portals* the following are the most relevant *protocols* and standards:

- ***Session Initiated Protocol (SIP)*** – The initial proposal developed for *VoIP* to define the technical parameters and details required for peer-to-peer session management:

    - **User location** – *IP* address, etc.
    - **User availability** – the willingness of the called party to communicate, with whom, when, and how
    - **Endpoint capabilities** – determination of the media types, media parameters (*Codecs / compression algorithms*, etc. – reference *H.323* below) and applications. These parameters can be part of the *protocol* for negotiating capabilities between undetermined devices
    - **Session set-up** – alerting or ringing a device, establish media session parameters at both the called and calling parties

- **Session management** – including transfer and termination of sessions, modifying session parameters, and invoking services
  <u>Note</u>: The scope of *SIP* was initially restricted so requirements for many applications like conference management or functions required by service providers such as requiring end-point capabilities negotiation for example, are done with extensions to *SIP* messages or in conjunction with *H.323 / H.248*.

- ***SIP* for Instant Messaging and Presence Leveraging Extensions (SIMPLE)** – is an add-on to *SIP*, which may become the basis for a new *protocol* over the existing **XMPP (Extensible Messaging and Presence Protocol)** used for *Instant Messaging* and online *presence* detection.

- ***H.323*** – the precursor, and relatively mature multi-media *protocol* standard, to *SIP*. There are many functions not yet defined in *SIP*, such as negotiating capabilities and communications between devices based on compression, etc. The majority of *VoIP* networks interoperating with the *PSTN* today are using *H.323*.

- ***Host Media Processing (HMP)*** – performs media processing functions via the CPU without any special-purpose telephony hardware.

- ***Voice XML (VXML)* –** a revolution in voice response application development has occurred with the widespread adoption of *VXML*. It leverages web-derived standards, including *XML* and the three-tier web architecture of back-end, web / application server, and browser. It brings welcome standardization to voice response hardware and software, and enables application portability between different vendor's platforms. It also enables the use of widely available web development tools and methodologies.

- ***VoIP Standards*** (reference 2nd paper on *UTM© for VoIP Network Design, including WiFi & Mobility*)

- ***ENUM*** – numbers are used to identify ordinary phones, fax machines, pagers, data modems, eMail clients, text terminals for the hearing impaired, etc. A prospective caller may wish to discover which services and *protocols* are supported by the terminal named by a given telephone number. The caller may also require more information than just the telephone number to communicate with the terminal. The holder of an *E.164* number or device may wish to control what *Uniform Resource Identifiers (URI's)* are associated with that number.

- ***IP Multi-media Subsystem (IMS)*** – is a set of standards being advanced to move carriers closer to a converged network architecture that can support development, deployment and delivery of applications over an *IP* backbone including converged voice services across wireless, enterprise and *WiFi* access networks.

- ***Interactive Connectivity Establishment (ICE)* –** a proposed standard by Microsoft and Cisco to enable *VoIP* packets to transverse firewalls. It will be provided on the client and intended to be supported by the various firewall manufacturers.

Applications

***Communication Portal* –** Unifies voice and web-based communications and management by incorporating *presence* and permissions-based parameters and access to user-specific devices. Each user manages their availability and preferences for communications with and among other portal subscribers.

Voice Messaging – Legacy *Voice Processing Systems,* which are typically stand-alone server-based providing telephone answering, Automated Attendant, Voice Mail, etc.; are giving way to media and application processor servers (as discussed in the first paper on *UTM© for Communication Servers & Contact Distribution Systems*).

***Unified Messaging* –** Combines Voice Mail, fax and eMail under a single server management and control. Access to all message types is done via the phone or web client. eMail headers and text content can be relayed over the phone using *text-to-speech* technology, including attached file names.

***Instant Messaging (IM)* –** Is the ability to determine whether a colleague is connected to the Internet / Intranet and, if so, to exchange text messages with them in near real-time. Some services provide the ability to attach files.

Interactive Web Response (IWR) Systems have been deployed using *Hypertext Markup Language (HTML)* and *Extensible Markup Language (XML)* for customer specific data interaction via a web browser. Initially systems used the back-end host database integration of *IVR* systems to pass customer-specific data to the *IWR* system via a *Common Gateway Interface (CGI)* script. This replicated the same functions as the *IVR* system. However, with the emergence of *Voice XML* this function reversed so the *IWR* system feeds back to the *IVR* system, since the *IWR* application can be more elaborate via the *graphical user interface (GUI)* and the *IVR* system functions as a subset.

Interactive Voice Response (IVR) Systems have been deployed for quite some time for customer specific data interaction via a phone with optional *Text-to-Speech (TTS)* and *Automated Speech Recognition (ASR)* more commonly referred to as speech recognition. An enormous number of different speech self-service applications have been successfully deployed. They include customer service applications in financial services, travel, catalog, and telecom. Many internal enterprise applications are also common, such as password reset and human resource benefits enrollment.

Automated Response and *Media Processing Systems* can be combined to include the following point solutions under a single server (including mirrored and *carrier-grade servers*) as opposed to having separate systems in multiple locations:

- **Traditional Telephony and Web Communications Applications being combined:**
    - Telephone self-service [*Interactive Voice Response (IVR) and Interactive Fax Response (IFR)*]
    - Web self-service [*Interactive Web Response (IWR)*]
    - Auto-attendant (*ASR* call director) / Voice Mail
    - *Audiotex*
    - Fax
    - eMail
    - *Unified Messaging*
    - Web callback
    - *Instant Messaging (IM)*, etc.

The following table, which is a progression from the previous paper on *UTM© for Self-service & Automated Response Systems*, provides a cross section of ***Communication Portals*** (Voice & Web Integration), ***Multi-media Application Processing*** and ***Cross-modal*** telecommunications relative to degrees of automation and timing for communication to take place between or among the various parties.

Computer-Telephony / Web Integration, Multi-media Application Processing & Cross-modal Telecommunications		
	Real-time communications	**Store-and-forward / retrieval**
Fully Automated Applications (Person-to-business)	Automated Attendant, *Interactive Voice Response (IVR) / Interactive Fax Response (IFR), Interactive Web Response (IWR), Knowledge Base* / key word search / *Natural Language Processing*, text / eMail / Fax / Multi-media Response	Voice / Video Mail, *Audiotex*, fax back, fax-on-demand, fax distribution (standard documents & forms), eMail
Semi-automated Applications (Subsequent to person-to-business application on transfer to agent)	Voice with optional screen / data pops to agent desktop (*CTI*), Web text chat, Web callback, Web co-browsing, Web page push to browser	Voice / Video Mail reply, eMail reply (Automated reply and / or person)
Manual Applications (Direct person-to-person or person-to-group)	Voice */ Instant Messaging (IM) /* Video */ Short Message Service (SMS)*	Voice / Video Mail / eMail personal reply

Figure 2

Telecommunication Modality

Real-time and store-and-forward (mail) telecommunication applications are being mediated through the use of ***presence*** (a feature of *SIP*) and ***multi-modal / trans-modal*** communications.

Modality **= Device / User Interface / Application / Media**		
Environmental	**Device (node)**	**User Interface / Application / Media**
Desktop	• Workstation / PC / thin client	• *Graphical User Interface (GUI)* for web integrated / *unified messaging* with optional soft-phone (voice, video, graphics, text, *ASR*, *TTS*)
Roaming	• Laptop	• *Graphical User Interface (GUI)* for web integrated / *unified messaging* with optional soft-phone (voice, video, graphics, text, *ASR*, *TTS*)
	• Wireless Phone for *VoIP* over *WiFi*; Cell Phone; and dual-mode (*WiFi* & Cellular) phones and PDA's	• Voice User Interface / and *WML GUI* on PDA implementation for web integrated / *unified messaging* (voice, video, graphics, text, *ASR*, *TTS*)
Mobile	• Laptop	• *Graphical User Interface (GUI)* for web / integrated / *unified messaging* with optional soft-phone (voice, video, graphics, text, *ASR*, *TTS*)
	• Personal Digital Assistant (PDA)	• Voice User Interface / and *WML GUI* on PDA implementation for web integrated / *unified messaging* (voice, video, graphics, text, *ASR*, *TTS*)
	• Cell Phone	• Primarily Voice / Touch-tone User Interface with optional *ASR*, *TTS*

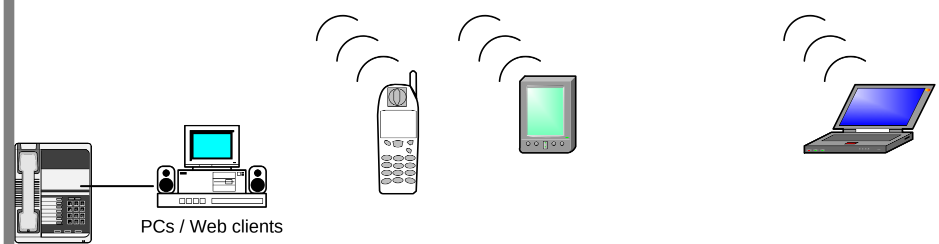
Figure 3

Configurations

Today's systems utilize open standards providing *Modular Communications Platform (MCP)* architecture. *Messaging & Communication Portals* can now be included within current generation *IP Communication / Media and Application Server* configurations as shown below in Figure 4.

Next-Generation Call and Multi-media Processing System

Call Processing **and** ***IP Multi-media Services (IMS) and Applications***
(In conjunction with Web and Smart phone / mobile clients)



Standard ***API's and Web Services***

Application Generator **Toolkit and Database Interaction via** ***IP***
(including the new *CSTA* specification for *CTI-type functions* with *SIP*)

Host Media Processing (HMP)

Operating System *(including hardened or Carrier-Grade)*

PCI, *Compact PCI*, or
Advanced Telecom Computing Architecture (Advanced TCA)
Media and Application Processor Servers and Blades
[No specialized *Digital Signal Processor (DSP)*-based cards required for *HMP*]

Figure 4

Working from the lower layers of the above configuration, the following provides further description:

PCI, Compact PCI, and Advanced Telecom Computing Architecture (Advanced TCA) – initiatives of PICMG (PCI Industrial Computer Manufacturers Group) that is a consortium of over 450 companies who collaboratively develop open specifications for high performance telecommunications and industrial computing applications and provide high-availability. *Advanced TCA* is an open industry specification designed the requirements for current and next-generation carrier-grade communications equipment primarily based on *IP*.

Operating Systems – include Linux and Windows. Carrier-grade versions are hardened to ensure against security attacks.

Host Media Processing (HMP) – With the increased performance of general-purpose processors, voice processing that once required dedicated digital signal processors (DSP's) can now be performed with software-only solutions on standard servers running Windows or Linux. *HMP* performs media processing functions without any special-purpose telephony hardware.

***Application Generator* Toolkit and Database Interaction via *IP* –** First generation systems relied on cryptic scripting languages. Some best-of-breed systems incorporated drag-and-drop application generators based on object-oriented programming. Few vendors with legacy systems were able to offer a common end-to-end development following the entire call process. With *IP*, all of that is changing since telecom vendors are now shifting to modular platforms using *IP*, and to embrace *web services* and common development environments to stay competitive. Moreover, with major standards converging around *IP* communications including *CSTA* and *SIP*, end-to-end application development has taken on a whole new dimension.

Standard *Application Program Interfaces (API's) and Web Services* – *API's* have been used for quite some time, including with first generation client-server architectures to enable a programmer to make requests from the operating system or another program. *Web Services* are a third-generation services-oriented, component-based application architecture using standard Internet *protocols* (*XML*, *Voice XML*, etc.) to make requests of other web applications. The client does not need to be a browser, but can be any type of Internet device, including a PDA or cell phone using *Wireless Markup Language (WML)*. Machine-to-machine transactions are done using *Simple Object Access Protocol (SOAP)*.

The following diagram (Figure 5) depicts a *Communication Portal Systems / Applications [based on Host Media Processing (HMP)]* using *Voice XML* in a *TDM* (legacy systems below the dashed line) and *IP* environment (above the dashed line).

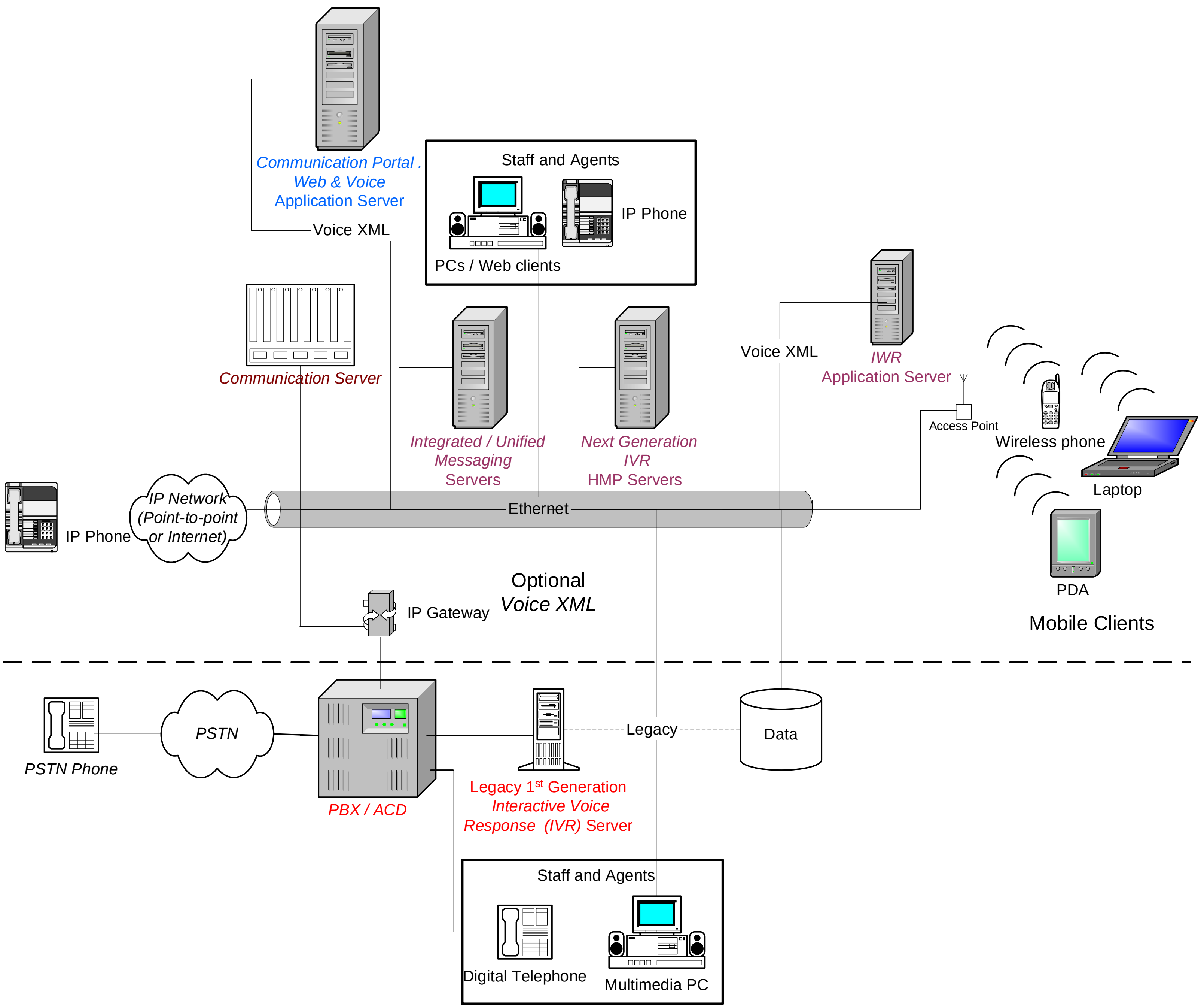


Figure 5

Normal configurations for *Voice XML*-based automated response systems conform to the web-based three-tier architecture; the application and associated business logic reside on a web server that serves *VXML* pages to “voice browsers”, also known as “voice *gateways*”. Unlike web-browsers, which are located with the user, voice browsers are normally housed in data centers, and users access them via telephone. The web servers connect to back-end data systems in exactly the same way as for web sites. In most cases, the same back-end data interfaces can be used for both web and voice response applications.

Next Steps

- Develop long term and short term strategy for going beyond traditional telephony, messaging and self-service point solutions
- Re-evaluate your current telecom model vs. a unified model for your particular enterprise and environment, including devices, user interfaces, applications and media for internal and external telecommunications
- Identify opportunities, impediments and solutions

Glossary

Advanced Telecommunication Architecture (ATCA) – is an industry initiative to provide standardized platform architecture for carrier-grade *IP* telecommunication applications, with support for carrier-grade features such as NEBS (Network Equipment-Building Systems) and 99.999% availability.

Access Point (AP) – a station that transmits and receives data and connects users to other users within the network and also can serve as the point of interconnection between the Wireless Local-Area Network (WLAN) and a fixed wire network.

Audiotex – *a* voice information bulletin board application that enables telephone users to hear pre-recorded information from a *voice processing system*.

Application Generator – a *graphical user interface (GUI)* using drag-and-drop objects to build an application and generate the required program code to run the application.

Application Processing System – provides operation and management / administrative features for various multi-media applications, which in the context of telecommunication architecture can reside on the same or a separate server from the media processing system that provides access to the required media resources.

Application Program Interface (API) – is a specific method prescribed by a computer operating system or by an application program by which a programmer writing an application program can make requests of the operating system or another application.

Automatic Call Distribution (ACD) – is an option that can integrate to most telephone systems which processes incoming calls to groups of agents. The ACD determines which group to transfer to, and if all agents are busy, plays a message and places the caller on hold and queues the call in sequence for the next available agent in that group. The system also provides management reports and information such as number of callers on hold, average hold time, etc.

Automated Speech Recognition (ASR) – is an option for a *voice processing* or *Interactive Voice Response (IVR)* system to enable input using voice. Types of ASR include speaker-dependent and independent; continuous [*Natural Language Processing (NLP) or Statistical Language Model (SLM)*] or discreet (limited to individual words or numbers).

Blade – A blade is a thin circuit board, containing one or more CPU's (making it a server) typically for a specific application within a modular rack which is energy and cost efficient.

Carrier-grade Operating System – an operating system that contains specific attributes to maximize security and uptime.

Common Gateway Interface (CGI) – a standard method for passing specific user information between a web server application and a user.

Communication Portal – provides users and groups a view into the *presence* and *modality* of their colleagues. Each user manages how and when they can be reached, including customized contact rules. Communication portals can also include *Web* and *Voice Portals*.

Communication Servers – also known as soft-switches are platforms that use software on industry standard servers to perform voice and multi-media communications over *IP* and via the *PSTN* through *gateways or SIP* trunks*.*

Compact PCI – a standard to define CPU form factor and backplane connectors

Computer Supported Telecommunication Applications (CSTA) – a standard which combines computing and switching *(CTI)* applications which includes a set of profiles that are especially tailored for CSTA applications that control and monitor an endpoint device (a *SIP* device, for example).

Computer Computer-Telephony (CTI) – Controlling the telephone system either from the client desktop (first party call control) or via a server (third party call control). CTI enables *voice processing* applications to monitor the status of telephone station sets and their availability within the switch. Call control functions include: answer call, make call / dial a number, release, hold, park, pick-up, call transfer (blind or supervised), conference, etc. Additionally, caller information such as *calling line identification (CLID)* can be passed through the telephone switch to a specified *voice processing* application. Commonly, it involves coordinating the transfer of a call in conjunction with sending screen pops to a desktop PC or thin client.

Cross-modal – the ability to retrieve, forward or reply to eMail or text messages with a Voice User Interface.

Customer Relationship Management (CRM) – a set of off-the-shelf or custom applications used to maintain customer information across the enterprise instead of disparate silos of information within departments. It also relates to best practices for improving customer satisfaction, retention and expansion.

Device Driver – is a program that controls a particular type of device that is attached to a computer. There are device drivers for voice cards, printers, displays, disk drives, voice cards and so on.

Digital Signal Processor – a CPU that performs specialized tasks, including voice and multi-media processing at a very fast rate compared to general-purpose CPU's. Essentially these chips process analog signals and convert them to digital form to facilitate analysis and manipulation.

Domain Name System (DNS) – a domain name is an Internet address, which are maintained by servers geographically dispersed throughout the Internet.

E.164 – is an international numbering plan, originally developed by the International Telecommunication Union (ITU) for public telephone systems in which each assigned number contains a country code (CC), a national destination code (NDC), and a subscriber number (SN). There can be up to 15 digits in an E.164 number. The ITU and the Internet Engineering Task Force (IETF) are currently working on a new plan called *ENUM* that will expand E.164 to encompass both traditional analog phones and digital devices, including computers and other devices on the Internet. All types of communications devices -- including analog phones and fax machines, digital phones, wireless (cellular) phones, pagers, digital modems, digital video terminals, and *VoIP* devices -- will have unique E.164 addresses with direct dialing possible from any device to any other.

Enterprise Application Integration (EAI) – is the integration of disparate or "best-of-breed" systems within the enterprise. For example, combining data or transactions that occur within a *Customer Relationship Management* application such as order entry with back-end applications such as inventory management.

Enterprise Resource Planning (ERP) – is a system that provides back-office applications including finance, human resources, inventory management, etc.

ENUM – is a standard adopted by the Internet Engineering Task Force (IETF) that uses the *Domain Name System (DNS)* to map telephone numbers to Web addresses or *Uniform Resource Locators (URL's)*. The goal of the ENUM standard is to provide a single number to replace the multiple numbers and addresses for an individual's home phone, business phone, fax, cell phone, and e-mail.

Extensible Markup Language (XML) – specifies content variables in terms of what data is being described. For example, the word "accountbal" placed within markup tags is a way to indicate that the data that follows is an account balance.

Gateway – a network point that acts as an entrance to another network and provides *protocol* conversion.

Graphical User Interface (GUI) – navigation of a computer screen using graphics (as opposed to text) and a pointing / selection device such as a mouse or touch-screen.

H.248 – is a recommendation to provide a single standard for the control of *gateway* devices in multi-media *packet* transmissions to allow calls to connect from a *LAN* to a *Public Switched Telephone Network* (*PSTN*).

H.323 – describes how multi-media communications occur between devices, network equipment and services.

Host Media Processing (HMP) –performs media processing functions of a communications application without any special-purpose telephony hardware. In the HMP model, toll-grade audio is transported as *VoIP* through the Ethernet network interface and all media processing occurs in the host CPU.

Hyper-text Markup Language (HTML) – is the standard language for publishing hypertext on the World Wide Web. It is a non-proprietary format that can be created and processed by a wide range of tools, from simple plain text editors - you type it in from scratch- to sophisticated authoring tools. HTML uses tags such as <h1> and </h1> to structure text into headings, paragraphs, lists, hypertext links etc. Its successor for current and future document types and modules is XHTML, which is a reformulation of HTML using *XML*.

Information Technology (IT) –is a term for encompassing both telephony and computer technology.

Instant Messaging (IM) – Is the ability to determine whether a colleague is connected to the Internet / Intranet and, if so, to exchange text messages with them in near real-time. Some services provide the ability to attach files.

Interactive Connectivity Establishment (ICE) – is a proposed standard by Microsoft and Cisco to enable *VoIP* packets to transverse firewalls. It will be provided on the client and intended to be supported by the various firewall manufacturers.

Interactive Fax Response (IFR) – similar to *IVR* below, but incorporates user-specific data in fax form (as opposed to voice) as a response to an automated request.

Interactive Voice Response (IVR) – enables the use of telephones or fax machines to effectively be used as data terminals for access to user-specific database information via a tone dialing keypad for navigation through menu prompts and entering data. *Automated Speech Recognition* can optionally be used for input. Output is via voice response using concatenated pre-recorded voice files, computer-generated text-to-speech, or fax [*Interactive Fax Response (IFR)*].

Interactive Web Response (IWR) – extending the functionality of *IVR* to the Web by leveraging the database access communications that is already in place with *IVR*. Additional functionality can be added using multi-media tools and content. Conversely, *IWR* can provide database information to the *IVR* system via *Voice XML.*

Internet Protocol (IP) – is the method or by which data is sent from one computer to another on the Internet. Each computer has at least one unique IP address. IP is a connectionless *protocol*, which means that there is no continuing connection between the end points that are communicating. Packets arrive in the correct sequence because of *Transport Control Protocol (TCP)*, the connection-oriented *protocol* that keeps track of the packet sequence in a message. IP Version 4 is prevalent with IP Version 6 now being deployed.

IP Multimedia Systems (IMS) – is a set of standards from the Third Generation Partnership Project (3GPP) designed to extend legacy call control and management for Internet and enterprise network multi-media communications, including end-to-end *QoS* and the convergence of wireless and wire line networks.

IP Telephony – provides all of the features and functions found in legacy *PBX*'s in *IP-based Communication Servers.*

Knowledge-base (KB) System – is an application for the dissemination of information, generally online, but can be extended to a *voice user interface (VUI)* using *Automated Speech Recognition (ASR)*. Information access can be automated using search engines, parsing eMails or by systems using artificial intelligent processes and a variety of media by users and / or agents in a contact center.

Media Processing Systems – interactive information systems comprised of computer-based hardware and software that integrate to *communication* servers / business telephone systems, directly to the Public Switched Telephone Network (*PSTN*), and / or data networks including the Internet for multi-media applications.

Modality – is the user interface as enabled by the device, context, and associated media used to communicate with people. This includes visual and speech interfaces, text and voice content, desktop and handheld devices, wired and wireless network connections, real-time and asynchronous exchanges, one-way vs. two-way messaging, person-to-person and group conferencing, application information delivery, and, finally, business vs. personal contacts,

Modular Communication Platform (MCP) – are industry standards-based communications infrastructure platforms and building blocks.

Multi-media Services – various media and combinations of media including voice [*Voice-over IP (VoIP)]*, audio, video, text, fax, eMail, etc. used within telecommunication systems and applications.

Multi-modal Communications – The ability for a user to contact and communicate with people or automated business processes in a choice of interface and content modalities. These modalities include any combination of the following:

- Visual text or Speech (voice)
- Conversational voice or messaging
- Real-time or asynchronous messaging
- Wired or wireless communication devices
- Handheld or desktop devices

Both contact initiators and recipients should be able to use any device for communication activities, subject to the restrictions of their functional device capabilities.

Natural Language Processing (NLP) – the recognition and generation of natural human language by a computer.

Operating System (OS) – the fundamental program that enables devices to be connected and applications to run on a computer.

PCI – a standard to define CPU form factor and backplane connectors

Point-to-point IP Network – allows corporations to extend their own corporate network through private and secure tunnels over the public Internet vs. leased lines for wide-area communication. Also known as a *Virtual Private Network (VPN)*.

Presence / proximity awareness – is a technique within a type of application that makes it possible to locate and identify a computing or communications device wherever it might be located when it is connected to the network. The device's location could be determined via a Global Positioning System (GPS). A user's proximity can be automatically determined, such as through *Radio Frequency Identification (RFID)*, indicating a readiness or ability to communicate in real-time via the connected device.

Private Branch Exchange (PBX) – also Private Automatic Branch Exchange. A customer premise switch that provides control and connection between station (telephone) sets within an organization based on *TDM and circuit-switching,* and provides numerous features and functions such as hold, transfer, conference, etc.

Protocol – is a specific set of rules that end points in a telecommunication connection used to communicate.

Public Switched Telephone Network (PSTN) – connection-oriented world-wide telephone network for voice and data.

Session Initiated Protocol (SIP) – is an Internet Engineering Task Force (IETF) standard *protocol* for initiating an interactive user session that involves multi-media elements such as video, voice, text chat, etc.

Short Message Service (SMS) – is a service for sending messages of up to 160 characters (224 characters if using a 5-bit mode) to mobile phones that use Global System for Mobile (GSM) communication.

Simple Object Access Protocol (SOAP) – is a way for a program running in one kind of operating system to communicate with another using *web services*.

Statistical Language Model (SLM) – is a type of speech recognition grammar (a set of rules specifying which words and phrases will be recognized) constructed using statistical analyses of thousands of examples of real user spoken input. Distinct from conventional prescriptive grammars created by application developers based on a priority of what their expectations are of what users will say.

Text-to-speech (TTS) – converting text to computer-generated voice.

Time-division multiplex (TDM) – is a method of putting multiple data or voice streams in a single signal (circuit) based on timing.

Trans-modal Communications – is the ability for communicating parties to dynamically switch from one mode of communication *modality* to another, without while maintaining the context of the prior communication. This would enable users to move from an asynchronous message (text or voice) either an Instant Message (IM) exchange or a telephone or video connection. An IM contact could also be escalated to a full-duplex voice or video connection. In all cases, informational attachments would remain accessible for collaborative reference in the new *modality*.

Unified Contact Distributor (UCD©) – is an evolution from the traditional *Automatic Call Distributor (ACD)*, which distributes real-time voice calls to agent queues. In addition to distributing voice calls. UCD's also control and distribute multi-channel, multi-media telecommunications in the form of real-time and non real-time communications. These include applications such as Voice Messaging, Fax, *Telecommunication Device for the Deaf (TDD)*, eMail, Text Chat, Web call-back, *Voice-over-IP (VoIP)*, Conferencing / Collaboration, and elements of *Computer-telephony Integration (CTI)* and Customer Relationship Management (CRM), etc., which are distributed to users and agent groups defined by skill sets and other business rules. Users and agents also have control over various functions within these applications via a *Graphical User Interface (GUI)* on thick or thin (browser-based) workstations and mobile devices. It follows the Integrated vs. *Unified Messaging* model of integrating stand-alone servers or having a single server that unifies / synchronizes the communications in real-time. Registered © CollabGen Inc. / eTelecom Consultants 2004 TX 6-066-696

Unified Messaging – combining voice / fax and eMail under a single server control. Access to all message types is done via the phone or PC / workstation. eMail headers and text content can be relayed over the phone using *text-to-speech* technology.

***Unified Telecommunications Model* © –** planned convergence of telephony and web communication that migrates from *circuit-switched* to modern *packet-switched Internet Protocol (IP)* telecommunications systems and applications in contact centers and across the enterprise. The objective is to provide end-to-end monitoring, management, and reporting regardless of the telecommunications channel, media, application or location. Registered © CollabGen Inc. / eTelecom Consultants 2004 TX 6-066-696

Uniform Resource Identifier (URI) – is the way to identify any points of media content, whether it is a page of text, a video or sound clip, a still or animated image, or a program. The most common form is a *URL*.

Uniform Resource Locators (URL's) – the unique address for a web site or file that is accessible on the Internet.

Virtual Private Network (VPN) – is a way for an enterprise to use a public telecommunication infrastructure, including leased lines and the Internet, to provide remote offices or individual users with secure access to its network.

Voice-over-IP (VoIP) – Voice over a *packet-switched* network using *Internet Protocol (IP).*

Voice Portal – provides entry to a voice application processor using *Automated Speech Recognition (ASR)* or touch-tone, which can be used for *IVR* and / or messaging and extended to web pages, general computing and *media processing.*

Voice Processing System – interactive telephone information systems, comprised of computer-based hardware and software, that integrate to business telephone systems or directly to the *Public Switched Telephone Network (PSTN)* or *VoIP* Network. Callers (or call recipients) can automatically be connected to digitally encoded voice recordings (sound), other telephone devices, and computer databases - all of which can be combined to provide an unlimited variety of fully and semi-automated applications.

Voice Processing systems allow the use of touch-tone phones and fax machines to enable the following categories of applications:

1) **Call Processing**
    a. Automated Attendant [call routing and automatic call distribution (ACD)]
    b. Voice Mail
    c. Voice (Audiotex) / Fax Information Bulletin Boards
2) ***Interactive Voice Response (IVR)***
    a. Enter and retrieve any information stored in a computer database
    b. Consultation with an internal or external *Knowledge-base (KB)* System.

Voice XML (VXML) – enables interactive access to Web applications and data through the telephone using touch-tone or *Automated Speech Recognition (ASR)*.

Web Portal – provides entry to a web site typically through a web browser [but can also include a *Voice User Interface (VUI])* which can be used for general computing and *media processing.*

Web Services – third-generation services-oriented, component-based application architecture using standard Internet *protocols* (*XML*, *Voice XML*, etc.) used to make requests of other web applications. The client does not need to be a browser, but can be any type of Internet device, including a PDA or cell phone using *Wireless Markup Language (WML)*. Machine-to-machine transactions are done using *Simple Object Access Protocol (SOAP).*

Wireless Fidelity (WiFi) – is a wireless Local Area Network.

Wireless Markup Language (WML) – is part of the Wireless Application Protocol (WAP) standard which is a language that allows the text portions of Web pages to be presented on cellular phones and personal digital assistants (PDAs) via wireless connection.