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Unified Telecommunication Model ©
White Paper

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

Communication Servers and Contact Distribution Systems
TDM / IP Telephony, Media Processors and Automatic Call Distributors (ACD) / Unified Contact Distributors (UCD©)

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Abstract:

This generic paper is the first in a series on migrating toward a ***Unified Telecommunication Model*** **©** in the enterprise.

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 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 **Executive / consultant checkpoints** [SM] that may be particularly significant to your organization's effectiveness and may warrant further 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 consultants. Independent telecom planning, procurement assistance and project management is provided in conjunction with its subsidiary, eTelecom Consultants (www.4etel.com).

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I. Executive Overview

Scope

This paper provides a framework for reviewing your organization's voice and web communication platforms in the context of traditional *circuit-switched* and new *packet-switched* multi-media contact distribution systems.

Benefits

Communication Servers are being deployed in various configurations with or without *Private Branch Exchanges (PBX's)*. Each provides the same primary functions which include signaling; database services; session connection; and end-user device communication requirements. *Communication Servers*, however provide more efficient *packet-switched* network resources using *Internet Protocol (IP)* compared to the dedicated *circuit-switched resources* of traditional *PBX* equipment. Major telecom and network vendors have responded by offering hybrid versions of their *PBX's* and / or via *IP telephony gateways* and pure IP *Communication Servers*. Since all telecom vendors have adapted *IP telephony* in various ways, there is a migration path available for most organizations that would not necessarily require a fork-lift change.

Communication Servers include ***Media Processing*** with some or all of the following Web-based applications: E-mail; *Unified Messaging*; Text Chat; *Interactive Web Response (IWR); Voice-over-IP,* etc., and also integrates / unifies traditional telephony point solutions. Therefore applications such as Auto-attendant / Voice Mail; Fax; *Interactive Voice Response (IVR)*; *Automatic Call Distribution (ACD);* Skills-based routing, etc. can now be implemented and managed using browsers and thin-clients under a single (unified) environment for all media. Consolidating and combining these applications within a *Communication Server* provide significant cost and overhead savings:

- Infrastructure is simplified and technology can be centralized which reduces or eliminates the need for maintaining disparate systems at different locations, but includes the option for mirrored hot-back-up distributed systems for multi-sites

- Specialized voice cards and dedicated circuits are no longer needed for voice mail, *unified messaging* and *IVR* systems

- Applications are implemented or easily changed and extended using browser technology including drop-down menus, radio buttons, and check boxes

- *Computer-telephony Integration (CTI)* projects and programming requirements are reduced and implemented more quickly so technology efforts can be directed to improve effectiveness, and customer retention / expansion

- Contact Centers are able to overcome obstacles to become Profit Centers

- Systems can be scaled quickly to meet the needs of ad hoc campaigns and surges in activity

- Increased built-in flexibility for continuity of business / disaster recovery, including virtual sites via remote / home workers and outsourcing options

- Comprehensive end-to-end real-time and historical reporting and analytics
- Administrative and supervisory functions via web browsers
- Web multi-media standards, including those enabling toll-quality voice for the enterprise, provide richer and broader set of added capabilities such as ***presence*** for new and enhanced converged applications.

Processes

In the context of multi-channel / multi-media communications including contact centers, the following provides an illustration for transforming from a non-integrated model (Figure 1) to a *Unified Contact Distribution (UCD©)* model (Figure 2):

Disparate Contact Distribution Systems

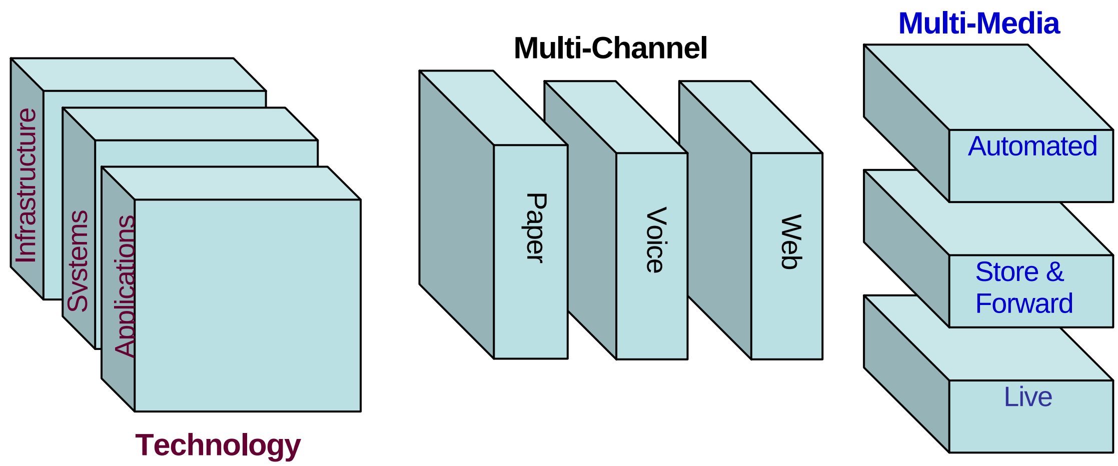

Figure 1

Unified Contact Distribution (UCD) ©

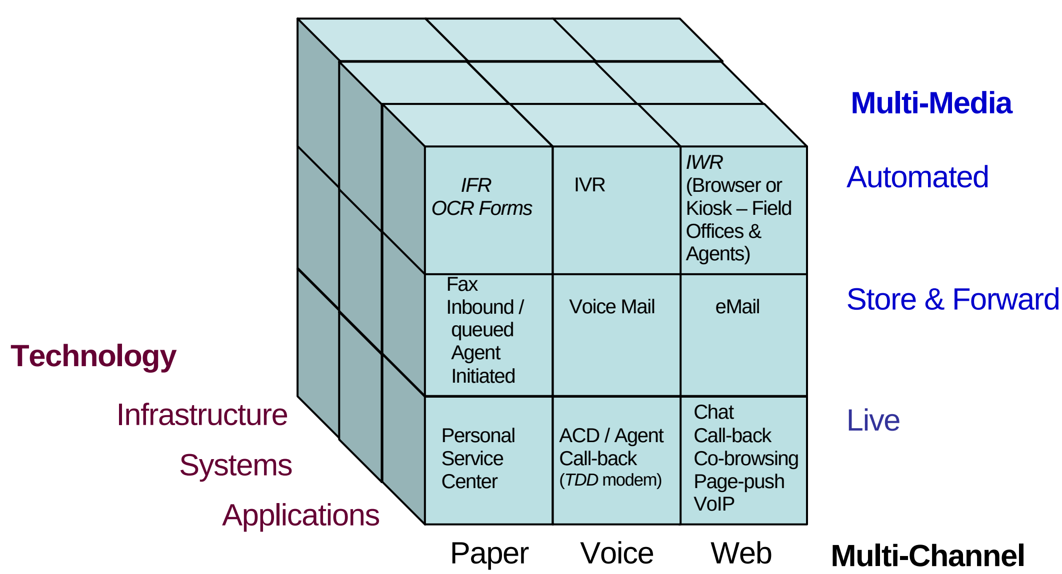

Figure 2

An example of proactively incorporating a *Unified Contact Distribution (UCD©)* model beyond concept is that paper can be reduced by converting correspondence to other media (scanned and attached in email for example) for more efficient distribution and tracking / reporting .

With converged voice and data networks in a contact centers, complex, costly and oftentimes lengthy integration efforts associated with *Computer-Telephony (CTI)* projects are minimized. Organizations can more readily establish contact routing rules based on information or attributes in databases to match agent skills with inquiries. Attributes can be determined dynamically with the incoming contact or by a variable in the database. Priority and skills-based routing can then be accomplished regardless of channel, media or location. In fact, knowledge workers outside the contact center could be brought into the communications stream based on *presence* and availability information through various media. *CTI* will be the subject of a future paper in this series.

New deployable standards have emerged including ***Session Initiation Protocol (SIP)*** and **SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)**, which is addressed in the following analysis section.

Technology Issues

In addition to migration issues for specific environments, there are considerations that need to be addressed for communication systems quality, reliability and security. More detail will be included in this and other papers in the series.

II. Analysis

Technological Domain

Communication Servers and Contact Distribution Systems are categories within Information Technology (IT) or Management Information Systems (MIS). Telecom traditionally required separate skill sets from data. With the convergence of voice and data networks these skills are becoming more complementary and are no longer mutually exclusive.

Deployment is occurring within enterprises, outsourcers and service providers. It is important to note that the technical and business requirements are considerably different for each. For example, inter-operability of different compression algorithms is much more complex for service providers that require universal / global access for their subscribers since the end-points can vary greatly as opposed to an inter-enterprise or an outsourced network where there are a definable set of devices and capabilities. Also, *Voice over IP (VoIP)* via the Internet has different variations of the issues found within the enterprise, which incorporate Service Level Agreements (SLAs) and effective *Quality of Service (QoS) parameters* such as *bandwidth, latency, jitter,* packet loss, security and reliability. Subsequent White Papers in this series will cover VoIP Network Design, including wireless and mobile applications.

Communication and Contact Distribution Servers encompass ***media processing systems*** and applications within their domain. These systems enable the use of ubiquitous tools including various forms of telephones, notification devices and web clients for real-time and store-and-forward communications and collaboration. *Communication Servers* as well as *voice* and *web portals* can enable users to manage their status and availability. *Presence* can be facilitated manually through a log-on process or automatically through schedulers and proximity-aware techniques such as using timeouts, *Radio-frequency Identification (RFID)* or even Global Positioning Systems (GPS).

Primary Enabling Technologies

- Media (voice, image, and video) digitization and compression
- Touch-tone recognition and generation
- Fax store-and-forward
- *Automated Speech Recognition (ASR)*
- Speech synthesis (converts text to speech such as reading email via audio device)
- *Radio-frequency Identification (RFID)*

Standards

There are various standards and *protocols* used for the media and technologies itemized above which need to be considered in detail when specifying associated systems and applications to ensure adaptability and inter-operability.

For *communication* and *media processing servers* 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** – how and the willingness of the called party to communicate and with whom

- **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

Note: 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. that are included in *H.323*. The vast majority of *VoIP* networks interoperating with the *PSTN* today are using *H.323* (since it is based on the current telecom standard ***Q.931*** for ***ISDN)*** or ***H.248 (Media Gateway Control Protocol)*** between *Communication Servers* and media *gateways.*

Applications

Communication Servers 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 and applications in multiple locations:

- **Traditional Telephony and Web Communications Applications being combined:**
  - Voice Conferencing
  - Auto-attendant / Voice Mail
  - *Audiotex*
  - Fax
  - *Interactive Voice Response (IVR)*
  - *Automatic Call Distribution* (*ACD*)
  - Announcements and music-on-hold
  - Skills-based routing
  - Web self-service
  - E-mail
  - *Unified Messaging*
  - Text Chat
  - Web call-back
  - *VoIP*
  - Web collaboration
  - Video Conferencing
  - Quality Monitoring and Recording
  - eLearning, etc.

Aside from the unnecessary overhead to support and manage stand-alone applications, it is important to note that many "front-door" applications do not adhere to best practices. Studies, including the use of video clips of users indicate a clear and compelling need for improvement in most enterprise self-service applications. Self-service and Automated Response Systems and some of the major principals for achieving best practices to satisfy and retain customers will be the subject of a future paper in this series.

Configurations

Design configurations are moving toward a combination of physical and logical functions over converged networks (a combination of the *PSTN* and managed *point-to-point* IP networks for voice). The following (Figure 3) provides an overview or progression from traditional telephony to incorporating IP Communication and Media Processors. Mirrored / hot standby *Communication Servers* can be incorporated at remote locations. In the event of a failure at one of the sites, the mirrored site(s) will provide automatic seamless continuity of business operations at all other locations. Outsourcing certain functions could also be part of this design even with IP *Communication Servers* from different vendors.

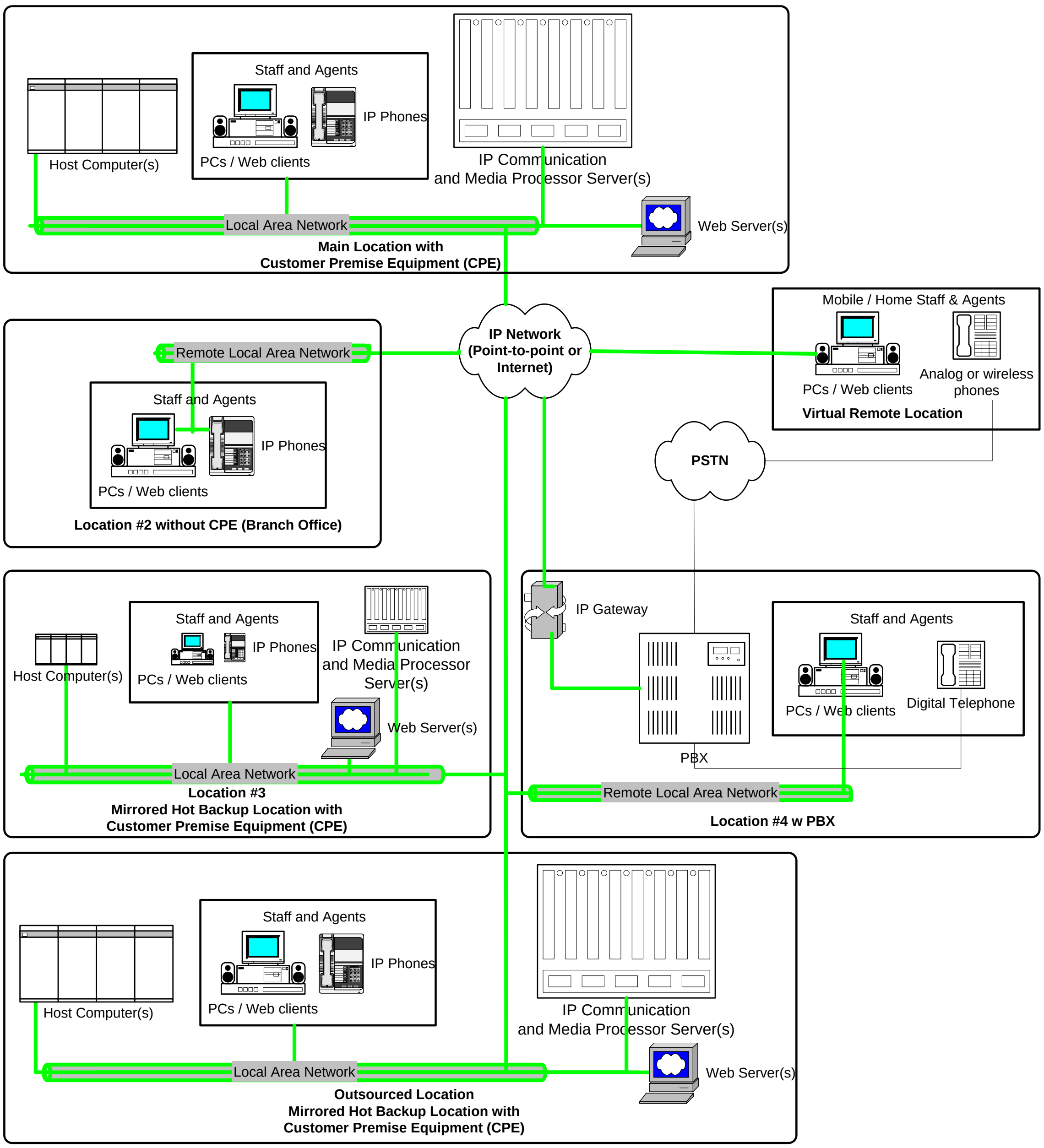

Figure 3

Next Steps

- Develop long term and short term strategy for going beyond traditional telephony and point solutions
- Re-evaluate your current telecom model vs. a unified model for your particular enterprise and environment
- Identify opportunities, impediments and solutions

Glossary

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

Automatic Call Distribution (ACD) – is *a*n 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* system to enable input using voice. Types of ASR include speaker-dependent and independent; continuous (*Natural Language Processing - NLP*) or discreet (limited to individual words or numbers).

Bandwidth – data transfer rate expressed in bits per second. G.711 is the international standard for encoding telephone audio on 64 kbps channel. It is a pulse code modulation (PCM) scheme operating at 8 kHz sample rate, with 8 bits per sample. So networks with a high data transfer rate can carry a large number of voice calls, especially in *packet-switched* networks using advanced *codecs*.

Circuit-switched – describes a network where a physical circuit (communications path) is dedicated for the duration of a call or media interaction with telephones or end-point devices (in contrast to *packet-switched*).

Codec – is an acronym for compression / decompression to reduce the size of files and can include analog-to-digital and digital-to-analog conversion.

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.*

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.

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.

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 multimedia communications occur between devices, network equipment and services.

Instant Messaging – Instant messaging provides immediate text chat capabilities with a pre-determined list of users, including the ability to see who is available to chat on your list.

Integrated Services Digital Network (ISDN) – is a set of standards for digital transmission over ordinary telephone copper wire as well as over other media through the *PSTN*. ISDN requires adapters at both ends of the transmission so your access provider also needs an ISDN adapter. There are two levels of service: the Basic Rate Interface (BRI) intended for the home and small enterprise, and the Primary Rate Interface (PRI), for larger users. The Basic Rate Interface consists of two 64 Kbps B-channels and one 16 Kbps D-channel providing up to 128 Kbps service. The Primary Rate Interface consists of 23 B-channels and one 64 Kbps D-channel in the United States or 30 B-channels and 1 D-channel in Europe. The D-channel carries control and signaling information.

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 Telephony – the use of *packet-switched* voice and data over enterprise networks or the world-wide web. Packet-switching for voice is reliant on network service levels and quality of service since the packets must arrive very close to real-time to ensure high quality voice reception.

Jitter –is the variation in the time between packets arriving caused by network congestion and other factors. A buffer can be used to mitigate jitter.

Latency – delays in packet reception causing choppy conversation in *VoIP.*

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

Multi-modal communications – combines speech with other channels of access such as keyboards and touch-screens to address business needs including mobility and extending customer service. In the context of computer-generated speech, multi-modal systems incorporate a means of adding non-verbal cues to speech (for example, animated gestures such as nodding or winking) to make the communication more clearly understood.

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

Optical Character Reader (OCR) – is the recognition of printed or written text by a computer using an optical scanner.

Packet-switched – describes the type of network in which relatively small units of data called packets are routed through a network based on the destination address contained within each packet which allows the same data path to be shared among many users in the network.

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.

Q.931 – is a signaling *protocol* for *Integrated Services Digital Network (ISDN)* communications that is used in *voice over IP (VoIP)*. The Q.931 *protocol* is involved in the setup and termination of connections.

Quality of Service (QoS) – the ability to manage or to some extent guarantee transmission rates, error rates, and other characteristics. For *VoIP* this includes *latency, jitter, packet loss*, etc.

Radio-frequency ID (RFID) – is a technology that incorporates the radio frequency (RF) portion of the electromagnetic spectrum to uniquely identify an object or individual. The advantage of RFID is that it does not require direct contact or line-of-sight scanning. An RFID system consists of an antenna, transceiver, and a transponder.

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

Telecommunication Device for the Deaf (TDD) – text based communications over the telephone network or other transmission facility using special modems based on the Baudot signaling method. They are usually acoustically coupled and typically operate at a relatively slow 300 baud or bps.

Text-to-speech – converting text to computer-generated voice.

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

Transport Control Protocol (TCP) – is a *protocol* used in conjunction with *IP* to manage individual data packets routing through networks including the Internet.

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 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 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

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 – enables entry or retrieval of information by speaking or pressing keys. The voice portal can respond with voice information or other form of media such as an email.

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 which can be used for general computing and *media processing.*

XMPP (Extensible Messaging and Presence Protocol) – is based on Extensible Markup Language (XML) and intended for *Instant Messaging (IM)* and online *presence* detection. It is server-based and facilitates near-real-time communications, primarily text.