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Unified Telecommunication Model © White Paper

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

Voice-over-IP Network Design

including WiFi & Mobility

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Abstract:

This generic paper is the second 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. Please contact **CollabGen** for copies of previous 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 (Generic)

- Scope
- Benefits
- Processes
- Technology Issues

II. Analysis (Generic)

- 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 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.

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

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.

Tim Connelly

Tim is an analyst and solution designer currently focused on mobility solutions including WiFi, VoIP and multi-media applications. With nearly 20 years of engineering experience in the computer and communications industry Tim brings depth of understanding of complex solutions for large and small organizations. He was in charge of developing wireless mobility solutions, including solutions to provide VoIP over WiFi for Enterprises with PBX integration and Seamless Roaming capabilities for a major telecom equipment manufacturer.

August / September 2005

I. Executive Overview

Scope

Media processing standards have emerged in conjunction with *Internet Protocol (IP)* as the dominant *protocol* for voice, data and video. Therefore applications can now be combined in new and more cost-effective ways in the context of a *Unified Telecommunication Model (UTM©)*. This paper will address ***UTM© for Voice-over-IP (VoIP)*** **Network Design** and an introduction to current *Voice-over-Wireless Local Area Network (VoWLAN)*, specifically ***VoIP-over-Wireless Fidelity*** **(*VoWiFi)*** and mobility. We will also discuss *PBX* / *Communication Server Integration* and *Seamless Roaming* capabilities. The broader scope of this paper includes all communication media-over-*IP* (voice / sound, image / fax, and video).

Benefits

VoIP provides significant cost and overhead savings:

- Can be consolidated on a single network infrastructure with data
- 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
- 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.
- 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
- Increased built-in flexibility for continuity of business / disaster recovery, including outsourcing options
- Comprehensive end-to-end real-time and historical reporting and analytics
- Administrative and supervisory functions via web browsers, including adds, moves and changes which eliminates the need to dispatch technicians in most cases
- Lower total cost of ownership

Processes - Enterprise Systems, Applications and Devices

In the first paper in the series, we covered *UTM© for Communication Servers* and Contact Distribution Systems. The following illustrates an overview of various enterprise systems, applications and devices, including those for desktop and mobile / remote users. Future papers will cover various topics shown below. This paper will focus on *VoIP* Network Design for Enterprises, including *VoWiFi* with *PBX* integration and *Seamless Roaming* capabilities:

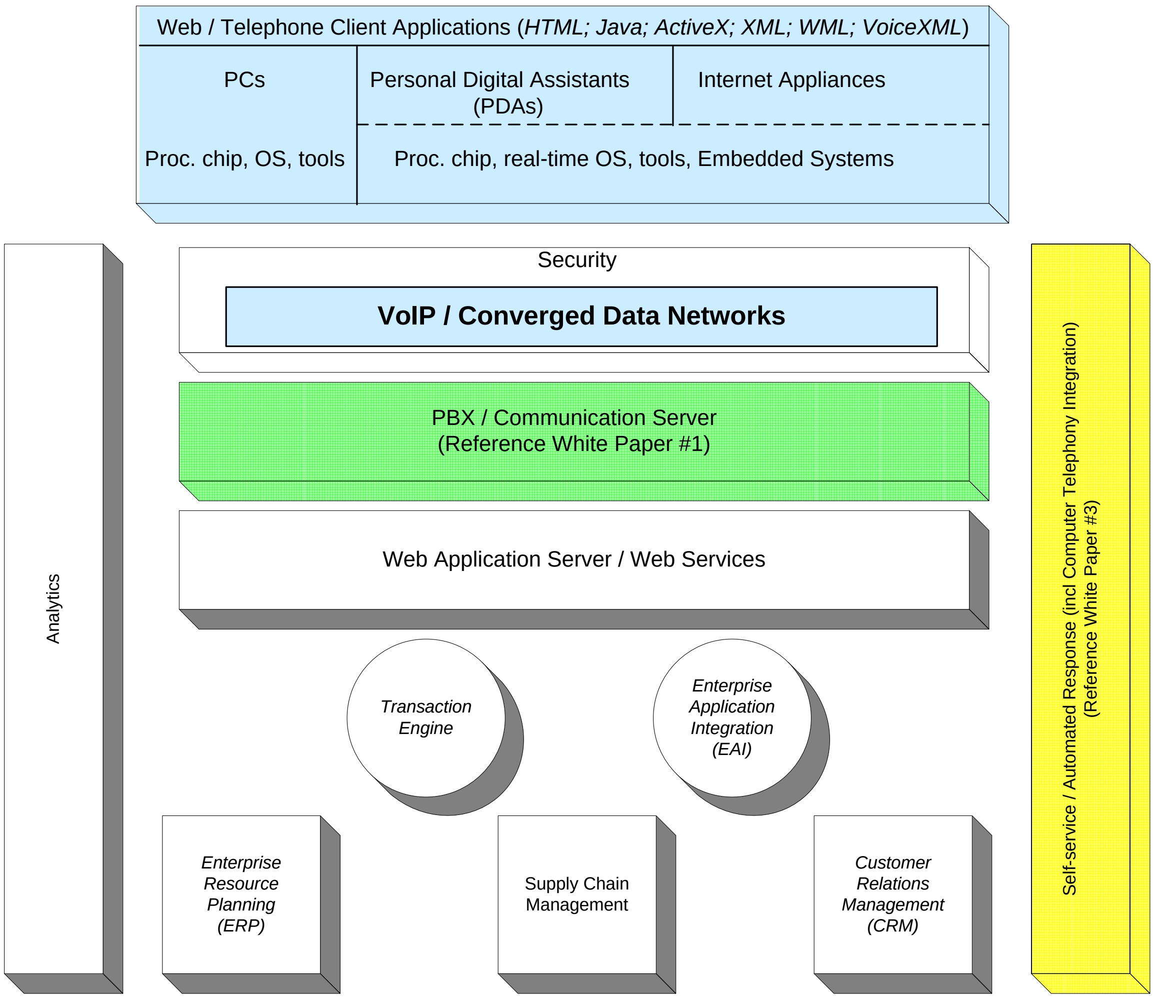

Figure 1

Technology Issues

In addition to integration issues for specific environments, there are considerations that need to be addressed for *VoIP* Network Design including quality, reliability and network security. More detail and solutions will be included in this and other papers in the series.

II. Analysis

Technological Domain

VoIP is a category within Information Technology (IT) or Management Information Systems (MIS). Telecom traditionally concentrated on *circuit-switched* networking and telephony, but now these circuits are part of the overall IT data networking domain.

VoIP expands, enhances and better integrates the use of ubiquitous tools including various forms of telephones, notification devices and web clients for real-time streaming, store-and-forward and collaborative communications. The difference can be dramatic, as seen with using *graphical user interface (GUI)* drag-and-drop controls for functions that were cumbersome with tone-dialing phones such as multi-party conferencing. Altogether, broader standards within *web services* in conjunction with status indicators, including *presence* information and *modality,* are opening up opportunities for new and dynamic applications.

Network Architectures – *VoIP* design and convergence:

VoIP – Quick Primer

Data networks conform to the ***Open Standards Interconnection (OSI)*** model developed by the International Organization for Standardization (ISO). In addition to the ISO, there are other bodies who contribute standards within the *OSI* model including the International Telecommunications Union (ITU), the Internet Engineering Task Force (IETF), American National Standards Institute (ANSI), Institute of Electrical and Electronics Engineers, Inc. (IEEE) and others. Respective contributions from these and other organizations that apply to *VoIP* will be addressed below and in the subsequent section on standards.

The *OSI* model is primarily for connectionless-oriented *packet* delivery [as contrasted with connection-oriented circuits via the *Public-switched Telephone Network (PSTN)*] and is based on 7 communication layers. These are: 1 – physical; 2 – data link; 3 – network; 4 – transport; 5 – session; 6 – presentation; and 7 – application. *VoIP* related standards and specifications are found primarily at layers 5 and 7, but rely on all of the layers for meeting design parameters critical to quality, reliability and network security required for voice communications and *PBX*-type features.

There are 3 major categories of data networks: *Local Area Network (LAN); Metropolitan Area Network (MAN);* and *Wide Area Network (WAN)*. IEEE provides most of the standards at the physical and data link (switching between computers on the network) layers for traversing these networks.

These networks are comprised of various combinations of the following types of *switches*:

Circuit *Switches* – originally developed in the late 1800's for the *Public Switched Telephone Network (PSTN)* for voice. These are connection-oriented in that they provide a continuous end-to-end path between users through various *switches* located in the network, currently using the centralized control of *Signaling System 7 (SS7)* and / or through control of a customer-premise *PBX*.

***Packet Switches* –** developed in 1970's for use by the Advanced Research Project Agency Network (ARPANet), which was the precursor to the commercial Internet. These are connectionless-oriented, where the *packets* are individually addressed and sent to the destination through various paths and re-assembled at the end user point. This method is much more efficient than *circuit-switches*. However, data doesn't mind delay but is intolerant to data loss or corruption, therefore the *IP protocol* and data networks were designed with that in mind: delays are fine as long as *packets* arrive in perfect condition, if corruption occurs then retransmit. Voice is the opposite; a small voice sample can be lost without the listener noticing so no retransmission is required. Not only because it wouldn't be noticed but because it would be too late. Delay is the major problem for voice and it must be minimized.

To address delay the industry developed *VoIP* techniques via the Internet intended to provide high *Quality of Service (QoS)*. These techniques eventually became industry standards. Basically these techniques prioritize voice and other delay sensitive *packets* so that they are forwarded ahead of normal data or other low priority traffic. Also, *bandwidth* can be reserved across the network for voice traffic. However it is important to note that these are still "best effort" attempts and do not guarantee a particular *latency* level. So even with these techniques special care must be taken in designing a network so that it can support *VoIP* effectively.

Frame *Switches* – developed in 1990's for internetworking between *LAN's*. This is connection-oriented, however is designed to tolerate *latency* and loss since *QoS* was not an issue for interoperability between *LAN's*. Although not as common as *VoIP*, if designed properly, *Frame Relay* can support voice and video [*Voice-over Frame Relay (VoFR)*].

Cell *Switches* – Cell is another term for *packet*, but is specific to Asynchronous Transfer Mode (ATM) technology. ATM is a connection-oriented switching technology designed primarily on hardware to accommodate faster speeds for all data types, including multi-media (voice / sound, image / fax, and video) with a guaranteed *QoS*. Therefore, *WAN /* ATM architectures are well suited for *VoIP* between sites.

Connecting two locations together for voice and multi-media communications has its challenges. Today many companies look to use the Internet as the carrier for *VoIP* between sites. This can be done with a variety of methods including *Virtual Private Networks (VPN's)* or *Point-to-point (PPP) IP Network* secure tunnels between the *routers* at each location. It is important to remember that these methods are primarily designed to provide a secure connection so that data cannot be compromised but does not provide a low *latency* connection. The Internet itself does not support *QoS* at this time so voice traffic will not be prioritized over data. Connections to the Internet are becoming more cost effective every day providing faster connections for lower cost. Many companies simply obtain large pipes to the Internet to compensate for this which can work effectively. Alternatives include using leased lines for *VoIP* traffic or using a *gateway* at each site utilizing the *PSTN*. If you need guaranteed *QoS* you need dedicated pipes. As discussed above ATM could be one of those dedicated *WAN* options.

The network layer includes *routers* and associated standards, including those that are vendor-specific. This includes ***Unicast*** and ***Multicast*** routing *protocols* thru *Internet Protocol* (IPv6).

Previously, networks consisted of *routers* and *bridges* which had very clear roles. *Bridges* connected local networks together making forwarding decisions based on layer 2 information in the *packet*. *Routers* were used to connect sites together making forwarding decisions based on layer 3 network information. As networks got larger *bridges* were replaced with *switches* which still made decisions at layer 2 but were much more efficient than *bridges*. They also reduced the size of broadcast domains which allowed very large networks to be constructed. However the lines between *switches* and *routers* have since become blurred due to the pursuit of more speed and the support of more applications such as voice. *Switches* now look at layer 3 information to perform functions like broadcast control and even forward *packets* based on layer 3 information. *Routers* now can perform like switches. A good example of this is *MPLS (Multi-protocol Label Switching) protocol*. With *MPLS* a tag is added to each *packet* which a *router* can use to make decisions quickly on where a *packet* goes without a full inspection of the layer 3 information. So today *switches* perform *router* functions and *routers* perform *switch* functions greatly blurring the lines between these devices.

From this point in the *OSI* reference model we move up through the transport layer, which includes *Transport Control Protocol (TCP)*, to the session and application layers where *VoIP* standards come into play.

Standards relevant to *VoIP*

There are various standards and *protocols* used for *media-over IP* and the enabling technologies itemized below which need to be considered in detail when specifying associated systems and applications to ensure adaptability and inter-operability.
For *VoIP and media processing* networks, the following are the most relevant general standards / *protocols,* moving from the lower layers of the *OSI* model on up:

- ***Multi-protocol Label Switching (MPLS)*** – designed to improve performance across the network. A tag is added to each *packet* which contains information on the *packet's* destination but it is at Layer 2 (switching) instead of Layer 3 (routing) which allows the *packet* to be forwarded much more quickly as it travels across the network.

- ***Unicast and Multicast Routing Protocols –*** these are layer 3 *protocols* and include IPv6 *Internet Protoco*l; *Resource Reservation Protocol (RSVP)* which enables channels or paths to be reserved in advance of the message being sent via the Internet and various *Multicast* (one source to many destinations) *Protocols* that are IP-based.

- ***Transmission Control Protocol (TCP)*** – provides layer 4 end-to-end connectionless-oriented *packet* delivery, including error recovery.

- ***VoIP-specific*** *protocols* begin at layer 5 and include:

    - **Real-time Control Protocol (RTCP)** – standard that specifies management of real-time transmission of multimedia data over either *Unicast* or *Multicast* network services.
    - **Real-time Conferencing** *(T.12x protocol* suite*)*
    - **Call Control** ***(H.245)*** – *protocol* for the transmission of call management and control signals in *packet*-based networks using *H.323* (see below) equipment.
    - **Call Setup** ***(Q.931 & H.225)***

    - Security and *compression* occur at the application layer 7. Network security will be the topic of a future paper in the *UTM©* series.

    - Layer 7 also includes:

- ***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

    <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 vast majority of *VoIP* networks interoperating with the *PSTN* today are using *H.323*

- **Cisco's Skinny Call Control Protocol (SCCP)** – defines a simple and easy to use client architecture for telephony applications in comparison with *H.323*. An *H.323* proxy server can be used to communicate with the SCCP telephone along with a proxy server for *H.225* & *H.245* signaling.

<u>For Wireless / *WiFi*:</u>

- **802.11a,b,g** – the well known *WiFi* variants
    - **802.11b** – 11 Mbps using the 2.4 GHz band
    - **802.11g** – 54 Mbps also at 2.4 GHz
    - **802.11a** – 54 Mbps using the 5.4 GHz band

- **802.11n –** an emerging standard for 100Mbps (and higher) *WiFi*. A critical component to the speed increase is "MIMO" technology; simply put MIMO (multiple inputs, multiple outputs) is multiple data streams within the same channel. There is significant controversy in the industry about MIMO, even in the definition of what it is.

- **802.11e** – standard based *QoS*
  Improvement to the *MAC* process provides layer 2 "over the air" prioritization between the client and *WiFi Access Point* as well as error correcting mechanisms to improve the performance of *VoIP* and other time sensitive applications.

- **802.11i** – security (described below)

- **802.11r** – Fast Roaming between *APs*
  Emerging standard to reduce client roam times from *AP* to *AP* from current levels, which can take 100ms to a full second or more to a consistent, reliable level targeted at below 25ms, perhaps less than 10ms.

- **802.3 –** Media Access Control – IEEE / ISO standard at the physical and data link layers for data encapsulation, error detection and recovery.

Primary *VoIP* Enabling Technologies

- Media (voice / sound, image / fax, and video) *digitization* and *compression*
- Touch-tone recognition and generation
- *Host Media Processing (HMP)*
- *Text-to-speech (T-T-S)*
- *Automated Speech Recognition (ASR)* and speech generation (voice response)
- Internet */ Web services*

Applications

VoIP 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 / Mobile Converged Communications Applications:**

  - Voice conferencing
  - Video Conferencing
  - *Unicast*
  - *Multicast*
  - *Multi-modal* applications, including Collaboration
  - *VoIP* phones using *WML* for custom applications
  - *VoIP* facilitates innovative ways to combine real-time communications with fully-automated and semi-automated applications in contact centers and across the enterprise (to be addressed in the next white paper on *UTM*© for Self-service & Automated Response Systems).

Configurations

The following provides a simplified overview of a *VoIP* implementation, including mobility with a traditional *PBX* integrated with *IP* multi-media */ Communication Server*: The existing phone system can be used intact with *VoIP* added to supplement it. In this illustration a *VoIP Gateway* is used to connect the *PBX* to the Ethernet / *IP* infrastructure. Call Control on the *IP* side is provided by a *Communication Server*. It accepts call requests and handles call control from both the *IP* clients and from the *PBX* via the *VoIP Gateway*. Calls can occur peer to peer between the *IP* clients, outbound to (or through) the *PBX* or inbound from the *PBX* or *PSTN*. Both wired and wireless *IP* clients are supported. Wired *IP* clients can include PCs or desktop *IP* phones. Wireless mobile *IP* clients can include laptop and notebook computers, PDAs and cordless style phones. *Access Points* provide the wireless connection between the network and the mobile clients.

Basic Overview

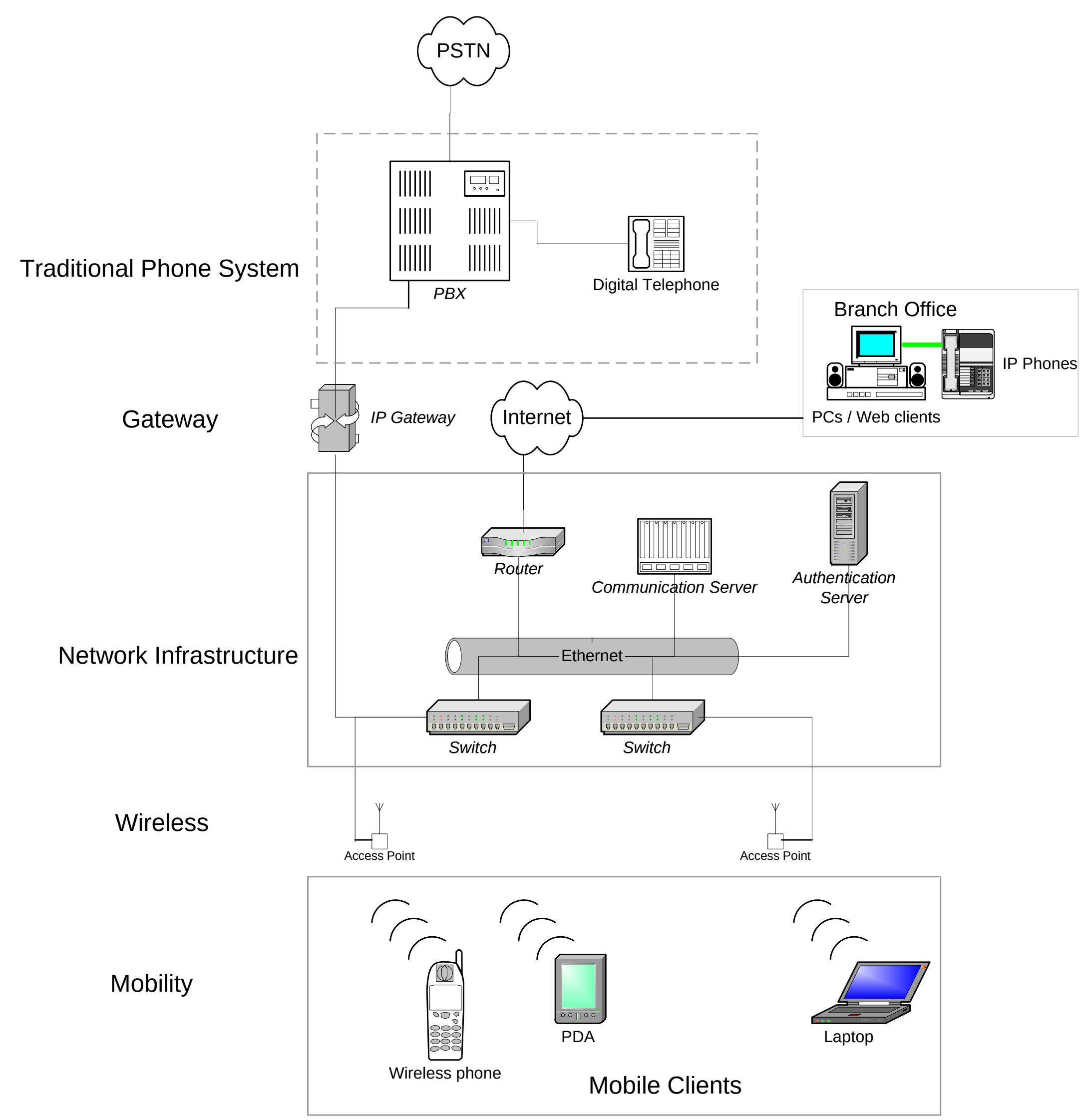

Figure 2

Design Considerations

Deployment of *VoIP* first requires an assessment of the existing network to provide a gap analysis for the work required to support voice. Typically an existing data network would not have been designed with voice in mind. The first area of concern is *latency* or delay. Data does not mind delay but is intolerant of errors; every *packet* must be received correctly even if it takes multiple retransmits, regardless of delay. Voice can tolerate a certain amount of *packet* loss but must have minimal delay. Enhancing the network to support *QoS* is generally required. The data load must be kept in mind as well, since the network is not changing from data only to voice only, it must support all the data requirements as well. This can be particularly challenging in mobile applications because the network and *access points* must continue to provide good voice service even under heavy data loads.

Reliability

Data networks must be reliable but since voice is still the most important communication method for business, the new voice network must be even more reliable than the existing data network. Traditional *PBX's* are famous for their ability to run for years without an outage and this level of service will be expected of the *VoIP* network as well. While *PBX's* are extremely reliable *IP* equipment has some distinct advantages. *IP* networks distribute functionality across multiple network devices and servers. This allows the opportunity to install multiple instances of equipment for load sharing and redundancy purposes. For example, *routers* and *switches* can run in parallel or be configured to "self-heal" around failed equipment. Multiple *Communication Servers* can be installed to provide additional redundancy. This is a great advantage for *IP* networks. In comparison it's not feasible to install two *PBX*'s in one company.

Another seemingly small detail can lend itself to improving reliability. Many *Access Points* and *IP* telephones can run on *power-over-Ethernet (PoE)*. These devices then get their power from a common wiring closet which probably also contains the *ethernet switches* and other equipment. Uninterruptible Power Supplies (UPS's) can be installed in these closets and since mobile client devices run on batteries the voice network could continue to operate even in the event of a facility power failure.

Integration to *PSTN*

Integration with the *PSTN* is an obvious requirement since the *VoIP* network cannot be an island. The *IP* voice *packets* must be converted to traditional *TDM* voice. This function is performed by a *VoIP Gateway* which might be a dedicated server or could be integrated into the *Communication Server*, a *router* or the *PBX*. Integration might also be handled by a service provider's network for a smaller installation.

911 / E911

An issue exists that has been getting much press lately on how to deal with emergency calls: since the phone is not hardwired from a specific port on a *PBX* how do you know where the person is? Most of the controversy has been related to residential use of *VoIP* but this still must be addressed for business *VoIP* systems. In reality this problem has been solved reasonably well as all the major vendors provide a method to define this in their system, however some analysis should be done to determine the best implementation. Smaller companies may want to route emergency calls directly to local emergency services with the business address as the location. Bigger organizations with large campuses could decide to route the calls to onsite security personnel or to forward these calls through the *PBX* so they are handled the same way as calls from traditional phones.

Usage Tracking and Billing

Management information for enterprises includes the information associated with *Call Detail Recording (CDR)* or end-to-end monitoring management and reporting for *VoIP*, including wireless communications.

Adding Mobility

VoIP-over-WLAN (*VoWLAN*) or *Voice-over-Wireless Fidelity* (*VoWiFi)* has the same concerns as wired *VoIP* with some additional considerations.

Security

The very nature of wireless technology means that the network can be reached physically from a distance. And that data between users and the network can be intercepted by others. Therefore it is imperative that appropriate security measures are taken to ensure that only authorized users gain access to the network and that *packets* transmitted are safe from eavesdroppers.

The basic process for authentication is useful to understand. The client (or Supplicant) requests connection to the network. The authenticator, in this case the *Access Point* intercepts this request and challenges the client for its credentials. The client replies sending the appropriate information, typically either User ID and Password or a certificate, which are encrypted. The authenticator forwards the credentials to the *Authentication Server* which verifies the credentials and replies to accept or reject the client. If accepted the Authenticator allows the client into the network. This two stage process can prevent the client from any access to any network resources until authentication is complete. The following provides a simple graphical view of authentication with a mobile client device:

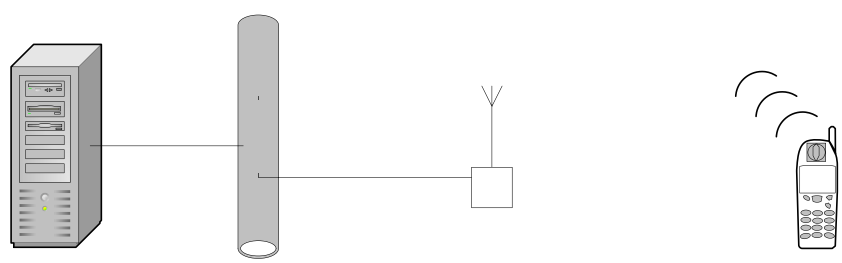

Figure 3

802.11i is an IEEE standard for security for *WiFi* networks. It was created due to the inadequacies of the previous standard known as Wired Equivalent Privacy (WEP). It greatly enhances data protection and client authentication typically leveraging Remote Authentication Dial-in User Service (RADIUS). It improves data integrity using Advanced Encryption Standard (AES), TKIP (Temporal Key Integrity Protocol), MIC (Message Integrity Check) and re-keying (dynamic or frequently changing encryption keys), as well as strong client authentication using EAP (Extensible Authentication Protocol) as 802.1x.

WPA (*WiFi* Protected Access) is an interim security proposal adopted by the *WiFi* Alliance based on 802.11i to bring a new standard to market quickly ahead of 802.11i. It also uses TKIP, MIC, EAP and dynamic encryption keys.

WPA2 is the *WiFi* Alliance version of the full 802.11i standard.

More network security detail will be addressed in future papers in the *UTM©* series.

Roaming

Implicit in a mobility solution is that client devices will be roaming from one *Access Point* to another. This effectively breaks the device's connection to the network and then re-establishes it on the new *AP*. When a client roams they must connect to the new *AP*, re-authenticate and re-start the voice *packet* stream. All this together can take a substantial amount of time, especially in terms of an active voice call. To solve this many vendors have developed custom solutions using their own equipment, additionally there is an emerging industry standard 802.11r intended to greatly reduce the time it takes for a client device to roam.

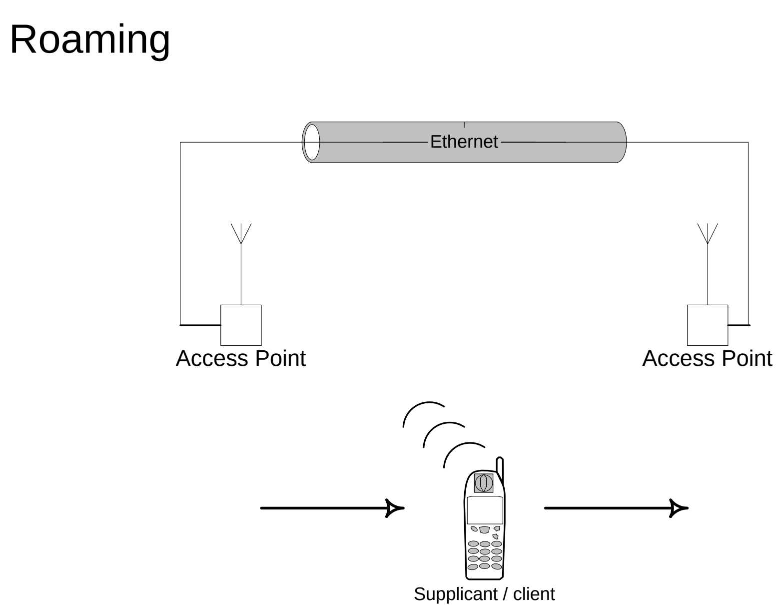

Figure 4

Communication Servers in conjunction with *AP's* can enable seamless handoff between *WiFi* and cellular networks, although single network number is an unresolved issue. Motorola announced in July 2005 that it is joining forces with a leading network vendor to deliver a **seamless enterprise mobility solution that will include *VoWiFi* telephony and cellular phone technologies**. By blending key features and benefits of mobile cellular, 802.11 wireless and enterprise fixed networks the combined solution will provide a single mobile communications device that bridges the physical and virtual-office environments to enable anytime, anywhere communications for mobile business professionals. A broader initiative, ***IP Multimedia 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.

Channel Management

There are many issues in deploying *VoIP*. One that is often misunderstood is channel management. *VoWiFi* uses several frequencies in the 2.4 GHz range which are referred to as “channels” numbered 1 to 11. When an *Access Point* is set to a particular channel it actually uses several adjoining frequencies. Therefore an *Access Point* on Channel 4 and another nearby on Channel 5 will actually interfere with each other. This is also referred to as “co-channel interference”. There are only 3 non-overlapping channels, 1, 6 and 11. Setting *AP’s* to these channels will prevent and interference with each other. As a side note, this is one of many problems caused by rogue *AP’s*, since it is likely that they can interfere with nearby *AP’s*.

WiFi Channel Management

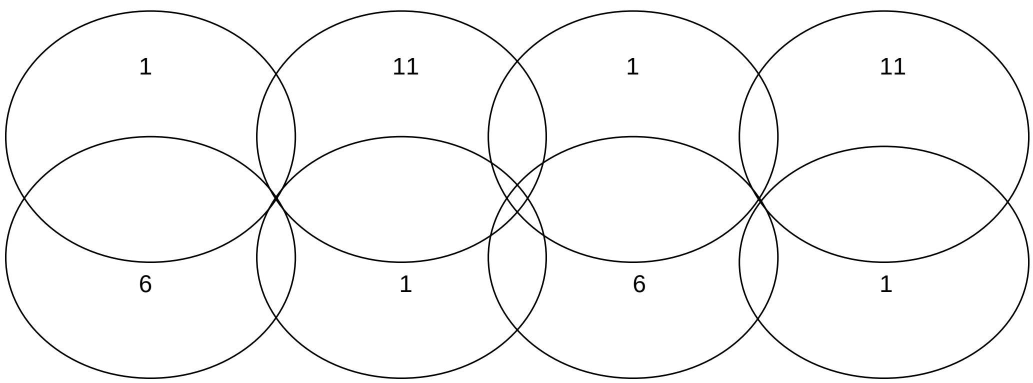

Figure 5

As you can see from the figure above this method is good but not perfect, the *AP’s* on channel 1 have some overlap. Typically it is impossible to avoid channel conflict completely, careful analysis must occur to avoid interference as much as possible. Also, this is a simple example; there are more sophisticated channel management strategies.

User density and distribution

In the same way as with a traditional voice network, you must determine how many out bound trunk lines are required. A mobile VoIP network design must take into account user density and distribution. This is because there is a limit to the number of simultaneous calls each Access Point can support. Typically 6 to 8 active calls can be supported on one Access Point. An over-subscription rate must be determined: i.e. 5 to 1 (total IP phones to current active calls) to decide how many APs are needed to support a given group of users. This must also take into account the call level for that group, a sales dept or call center will make many more calls than the engineering dept. The number of *APs* and the area of coverage (or "cell") of each *AP* will be determined partially from this analysis.

Next Steps

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

Glossary

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 *WLAN* and a fixed wire network.

ActiveX – is a set of tools provided by Microsoft that runs on Windows and Macintosh systems, which are essentially similar to *Java*.

Authentication Server – provides authentication, authorization and accounting for *VoWLAN* / *VoWiFi* implementations.

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*.

Bridge – connects multiple *LAN's* that use the same *protocol* to broadcast messages to all possible destinations, which is not feasible in larger networks – hence *routers* and *switches*.

Call Detail Recording (CDR) – file containing information about *VoIP* system usage such as the identities of sources and destinations (endpoints), the duration of each call, the total usage time in the billing period, etc.

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.*

Compression – using algorithms to reduce the size of files in a communications stream, including removing spaces and inserting codes for repeat characters, and other techniques.

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.

Digitization – converting analog signals to digital formats enabling *VoIP*. Advantages include more efficient real-time streaming such as voice or video, and store-and-forward applications such as voice 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.

Frame Relay – *packet-switched* networking, initially developed in the 1990's to support data internetworking among *Local Area Networks (LAN's)*.

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.225 – a part of *H.323* that provides a method for providing communications between end-points in a non-*QoS* environment.

H.245 – a part of *H.323* that provides control messages between end-points.

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.

Hypertext Markup Language (HTML) – is a formal recommendation by the World Wide Web Consortium (W3C), to which major browsers adhere, including Microsoft Internet Explorer and Netscape Navigator (although each has extended features which need to be considered for universal access). Current version is 4.0.

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.

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 info

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 – 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.

Java – a relatively simple language developed by Sun Microsystems designed to run on a centralized or distributed (via the Internet) platform-independent environment. It can be used to build small applications (applets) enabling users to interact with the web page.

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.*

Local Area Network (LAN) – a network for sharing resources and communications between users within a small geographic area, such as an office or other building.

Metropolitan Area Network (MAN) – a dispersed network within a geographical region. A MAN is between the geographical ranges of a *LAN* and *WAN*.

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.

Modality – speech, handwriting, keyboarding, pointing, graphics, video etc. used in multi-modal communications thru specific types of devices [phones, browsers, Personal Digital Assistants (PDA's), etc.] and associated applications.

Multi-modal communications – combines speech with other channels of access such as voice, keyboards and touch-screens to address business needs including mobility and extending customer service. Multi-modal communication applications include real-time streaming such as voice or video, store-and-forward applications such as eMail, and collaborative applications such as conferencing. 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 by an avatar) to make the communication more clearly understood.

Multicast – network communications between a single sender and a many receivers.

Open System Interconnection (OSI) – is a reference model for and from which vendors are able to develop detailed interfaces that in turn could become standards. OSI includes standards from other major standards bodies. Generally, OSI defines 7 layers; layers 1 to 3 define communications between operating systems, and layers 4 thru 7 define messages between users. OSI is a standard of the International Organization of Standards (ISO).

Packet / 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 (PPP) 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).*

Portal – user access thru a web and / or voice client to a web site or shared program. Depending on the nature of the application, this can be used to access general information or provide interaction for obtaining user-specific information such as via *Interactive Web Response (IWR)* and *Interactive Voice Response (IVR)* or other forms of *media processing*. In multi-modal telecom applications, status information can be changed in conjunction with scheduling or *presence*.

Power-over-Ethernet (PoE) – A means for providing power to a device, including *VoIP* phones, using the same cable lines used to deliver Ethernet using Ethernet Category 5 or higher data cabling.

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.

Router – resides at a point connecting 2 or more networks, included within a *gateway* or sometimes a *switch*, and determines the next network point to which a *packet* is to be forwarded.

Seamless Roaming – is the ability for a mobile device to move from one *Access Point (AP)* to another automatically while maintaining the communication. Voice requires this to be accomplished with very low latency between connections. Dual mode phones will support roaming between *VoWiFi* and Cellular networks.

Resource Reservation Protocol (RSVP) – is used by the host computer to request quality of service from the network through multiple gateways along the path of *packet* transmission in *Multicast* and some *Unicast* applications.

Signaling System 7 (SS7) – within the *PSTN*, a separate channel from the actual voice channel used to control and manage telephone calls, including various telephony features for wire line and wireless systems.

Switch – analyzes each *packet* or data unit and determines from a physical (Media Access Control – MAC) address which device a data unit is intended for and switches it to that device. The term is also used to refer to a *PBX*.

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.

T.12x protocol suite – a set of platform-independent *protocols* used to establish secure voice and multi-media conferences with various features including data communications, sharing applications, and white-boarding sessions.

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.

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

Unicast – network communications between a single sender and a single receiver.

Unified Telecommunications Model (UTM©) – 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-Frame Relay (VoFR) – Voice-over *packet-switched*, connection oriented *Frame Relay* network.

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

VoIP over Wireless Fidelity (VoWiFi) – Voice over *packet-switched* network using *IP* and *Wireless Fidelity* under the IEEE 802.11 set of *protocol* standards

VoIP over Wireless Local Area Network (VoWLAN) – is a set of technologies for sending voice over a wireless digital broadband *Local Area Network* to an authenticated wireless phone device, portable computer or Personal Digital Assistant (PDA).

Voice Portal – enables entry or retrieval of information by speaking or pressing keys from a tone-dialing phone. The voice portal can respond with voice information or other forms of information and *media processing* such as an email.

Web Portal – provides entry to a web site which can be used for general computing and *media processing.*

Web Services – are services and associated technologies, standards and applications made available from World-wide-web which can be applied to networking systems and applications, including *communication servers*, *media processing systems* and *VoIP*.

Wide Area Network (WAN) – is a widely geographically dispersed network, which can be private or leased, but typically includes either or both the *PSTN* and Internet.

Wireless Fidelity (WiFi) – the term applies to products that use a specification created by the WiFi Alliance that provides product certification and is part of the 802.11 standard. WiFi has gained wide acceptance in many enterprises and homes as an alternative to a wired *LAN*. Many airports, hotels, and other public facilities offer access to WiFi networks (also referred to as 'hot-spots').

Wireless Markup Language (WML) – part of a broader standard, Wireless Application Protocol (WAP), that allows the text from Web pages to be presented on cell phones and personal digital assistants (PDAs).

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.