UNIFIED COMMUNICATIONS

TECHNOLOGY OVERVIEW

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Unified Communications

Technology Overview

Introduction

Unified communications is a fast growing sector in the telecommunications industry, representing significant opportunities for new service offerings. This paper provides a high level overview of the unified communications industry and the technology behind it, written for a non-technical reader who needs to understand the basics. It may be of particular interest to investors, entrepreneurs or others looking to offer unified communications services.

Unified communications is an umbrella term for several related services: single number service for multiple phone lines, fax, email and messaging; access to voicemail through an email client; access to email through a telephone; managing telephone calls, conferences, etc. using a computer or web interface; and so on.

Several major market trends have provided the impetus for this market to grow. High penetration of mobile phones, combined with the high growth of messaging technologies such as email, voicemail, SMS and instant messaging, has led to a level of personal mobility and "connectedness" which enables business and consumer users to be available and in communication at any time, from any place. With this capability has come increased complexity for the end user, and unified communications represents an opportunity to reduce this complexity.

Technology advances and cost reductions in such areas as Voice over Internet, wireless data applications, Interactive Voice Response (IVR), subscriber management applications, and others have created the capability to capitalize on these trends in both the equipment and service provider areas. Unified communications combines these

existing technologies (many of which are themselves still maturing and advancing) to create a new technology sector which is highly competitive and rapidly evolving.

As a result, unified communications is highly complex from both a technology and market perspective. At this stage, there is no universally accepted definition of unified communications, and no established business model that has been successful on a large scale. The definition changes according to current technology capabilities (which vary from vendor to vendor) and perceived market opportunities and need.

Technology Environment

Voice and Data Networks

To understand the evolution of unified communications, which is the basis for single number services, merged voicemail/email messaging and a variety of related business opportunities, it is important to understand the larger context of voice and data communications networks. Voice and data networks have historically been very separate.

The most significant difference between traditional voice and data networks for our purposes is that voice networks connect terminal equipment (i.e. telephones) using a method called "circuit switching", whereas data networks connect terminals (i.e. computers) using "packet switching". Circuit switching uses a fixed, dedicated path through the network to make a connection – any data between two points (such as a digitized voice stream) uses the same reserved path. Because this path is established between a series of digital telephone "switches", the public telephone network is usually referred to as the "Public Switched Telephone Network", or PSTN.

In contrast, packet switching "chops up" a data stream into "packets", each of which can take any of a number of paths through the network before being reassembled at the destination into their original order. There is no fixed path any given packet must take – each packet is routed across the network based on the best path available at any given time. Packet switched networks have an equivalent to the digital telephone switches in the PSTN which are called "routers" or "packet switches" depending on the specific type of equipment used.

Voice is considered to be "delay sensitive" and "loss sensitive" network traffic, meaning that if traffic across the network is delayed more than about 100 milliseconds, or if more than about 2% of packets are lost while traversing the network, there is a noticeable loss of quality (static, choppy audio, echo and other problems). Circuit switching, because it uses a reserved, fixed path, isn't as susceptible to delay or loss, but is less efficient use of network resources than packet switching. Thus circuit switching has historically been preferable for voice or video network traffic. Data traffic, on the other hand, because it is not usually "real time", is not delay or loss sensitive. For example, if packets of an email message or file transfer arrive hundreds or thousands of milliseconds late or get lost in

transit and have to be retransmitted, there is no noticeable effect on service quality. Thus packet switching has always been preferable for data traffic.

Since the digitization of voice networks in the 1970's, one of the major themes of telecommunications has been that of *convergence*, the coming together of voice networks with data networks to create a single infrastructure capable of managing and delivering voice, video and data. More recently (say in the last 20 years), convergence has come to mean more than just using a single network for delivering voice and data services, and is often used in reference to entirely new multimedia services that deliver real time voice and video combined with multimedia content, data and transaction processing, man-to-machine communication, advanced computing and other intelligent services. Unified communications is one application of this capability, but there are hundreds of others.

There are two general models for delivering converged services like unified communications. The first, which is more practical for currently available business opportunities, takes into account that voice networks (i.e. telephones) and data networks (i.e. the Internet) are separate systems and technologies, and attempts to bridge or integrate these existing systems together to create new services. The second model is based on the eventual migration of all voice services to an Internet-based converged network, and is designed to work on a single infrastructure. At present, the market for the second model is largely limited to internal company communications, but is expected to eventually move into public communications services for businesses and consumers, replacing the existing telephone network.

Unified Messaging and Unified Communications

Once it is technically feasible and economically viable to integrate or converge voice and data networks, many new services become possible. One of the most popular of these services has been the ability for a user to access voice mail, email, faxes, pages, SMS messages or any other form of message via either a voice interface (i.e. the telephone) or via a computing interface. This is called "unified messaging". Unified messaging emerged in the 1990's with such companies as Glenayre and Comverse, who is still a major player in this industry today.

A "unified messaging" system has the following key defining components:

1. **Message Platform**: First, a unified messaging system replaces/modifies traditional voicemail and email storage with a converged storage and retrieval system capable of handling multiple media (e.g. email, fax, voice message, etc.). The most common generic term for this is the "message platform", and they are usually designed to either replace or modify existing email platforms (such as Microsoft Exchange, Lotus Mail, etc.)

2. **Communications Server**: Second, a unified messaging system allows "multi-modal" access, meaning the user can access voice and email messages either

through a computer or via a voice or keypad interface on a telephone. This multi-modal access is generally the heart of the unified messaging system, and has interfaces to both voice and data networks, with hardware and software on the platform needed for processing and conversion between formats. This device is sometimes called a "telephony server" because it connects the data network with the telephone system.

3. **Back Office**: Third, a unified messaging system has a computer-based "back-end" system consisting, at a minimum, of the following hardware/software:

    a. a user directory containing a record of each user, his/her various addresses/numbers, communication preferences, etc. (this database is usually based on a common directory format/protocol called "LDAP")
    b. a security server which authenticates each user (usually by a password), assigns rights and permissions, and generates usage/billing records. These are usually called "triple A" or "AAA" servers, for authentication, authorization and accounting (the most common type of security server uses a protocol called "RADIUS").
    c. a management application used to create and define new users, delete old users, change permissions, etc. These are usually icon-based graphic user interfaces custom built by the individual vendors. This is most commonly called a "provisioning" system.

    These back-end systems vary greatly in physical configuration, features and options. Some are integrated into the unified message platform itself, others are on specialized cards in the same chassis, and still others are completely separate boxes connected on the same local area network. All of these possible configurations have different advantages and disadvantages, and should be evaluated according to specific operational and service requirements during a formal vendor selection process.

"Unified communications" almost always includes a unified messaging function, but adds additional services to it. Unified communications is most commonly viewed as an evolution of unified messaging which includes real-time communication in addition to messaging. These real-time communication services include (but are not limited to):

1. Single number or "follow-me" service (a single number is used for multiple voice lines (e.g. home, office, mobile), as well as fax, and the number also serves as an email, paging and messaging address)

2. "Presence management" features (which allow users to set preferences as to how, when and where they should be communicated with, generally through a web based interface, as well as letting callers receive selected status information on the user)

3. "Call features" which are typically available with traditional voice services (such as call forward, call display, conferencing, call blocking, etc.)

4. Personal data services, such as managing contacts and personal calendars, database access, information services, etc.

These features may be built in to one of the unified messaging platforms, or they may be provided through separate devices on the network.

Unified Communications Technology

Network Environment

The diagrams below illustrate a simplified unified communications system. Figure 1 shows the end-to-end network, including the two public networks (PSTN and Internet) as well as the various communication devices any individual user may have (phones, faxes, computers, pagers, personal digital assistants, etc.). The unified communications system must be able to provide service to each of these devices, as well as coordinate between them.

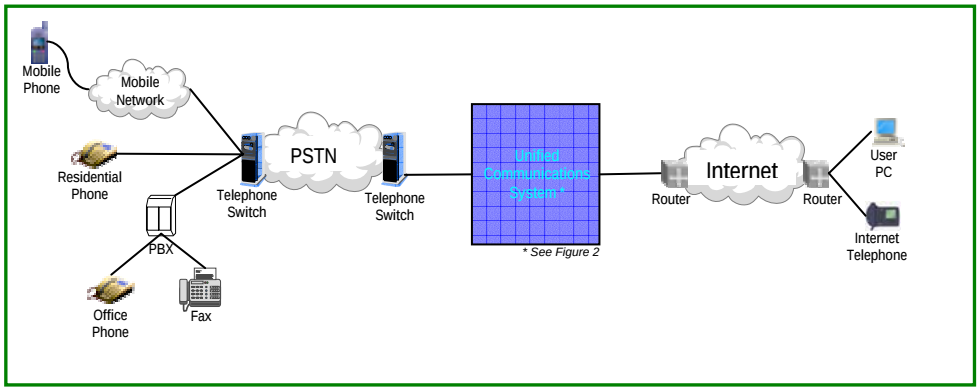

Figure 1

In many cases, the communications device in question may not be connected directly to the public PSTN or Internet. Computers, for example, may use a modem or high speed access technology like ADSL to connect to the Internet directly. However, most corporate computers are connected to internal company networks which connect to the Internet through a firewall or some other kind of gateway device. Similarly, many business phones and fax machines are connected to the PSTN through a "private branch exchange" or PBX.

In order for the Unified Communications System to connect to the PSTN and Internet, it is necessary to purchase connectivity from a network provider, generally the telephone company. A number of technologies can be used for this connection, but the most common is called a Primary Rate Interface or PRI line. Other alternatives, which vary

according to a number of factors, have names like T1, E1, T3, OC3, and so on. The generic term for all of these services is "leased line". A leased line can be used either for multiple telephone lines or for Internet access, depending on the network equipment to which the leased line is connected.

The unified communications system itself is illustrated in Figure 2. This is a simplified, generic view to illustrate the main platforms and functions of the system. A real-life implementation would vary according to things like the size of the system, the vendor(s) used, etc.

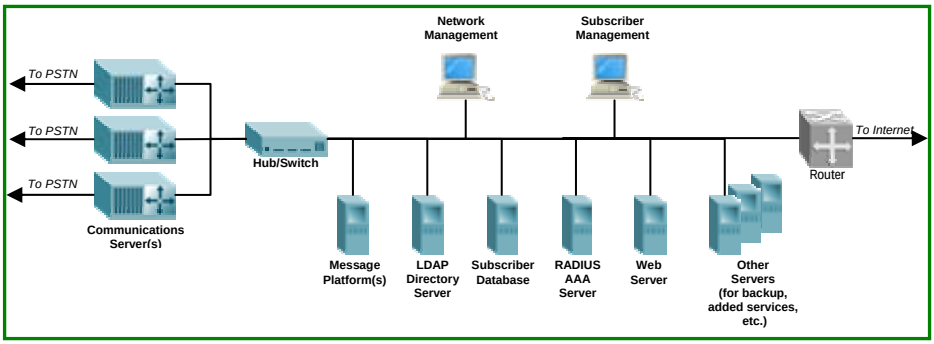

Figure 2

In general, the servers on the bottom of the diagram (message, directory, web, etc.) consist of software running on a generic Windows, Linux or UNIX server. The network and subscriber management consoles are usually mid to high end workstations running software applications supplied by the unified communications vendor, and in small networks the two functions can be combined onto one station (although this is usually not recommended for a number of reasons). Other devices shown (communications servers, hubs, routers, etc.) are specialized networking equipment.

A brief description of each of these components follows:

- **Communications Server**: This device is the heart of the unified communications system, and provides the interface between the telephone network and the data network. Functions include the ability to receive or originate PSTN calls and interactive voice response (i.e. voice prompts to direct the user to press number keys to execute various commands). Functions may also include text to speech conversion, speech recognition, and advanced call features (such as conference calls, call display, etc.). The communications server must have enough interfaces to the PSTN to support the peak number of simultaneous calls going through the system at any time. For example, if the service has 10,000 subscribers, and it has been determined that, at peak times, 1 out of 5 users is making or receiving a call, it would be necessary to have enough communication servers to support 2,000 calls. The most common type of PSTN interface is a PRI or T1 "port", with each port supporting 24 individual lines or connections, so to support 2,000 simultaneous calls 84 PRI interfaces would be required. If each communication

server had a capacity of 24 PRIs, a minimum of 4 communication servers would be required for the system.

- **Hub/Switch**: This is a device which acts as a "traffic director" on the local network to manage communications between all the various devices. It generally consists of a number of physical ports to which all of the other servers are connected. A hub is a low-end, low-capacity device, whereas a local area network switch (for example, a Cisco Catalyst 5000) is designed for larger networks with greater traffic. Depending on network configuration, the switch may also provide the function of the router (see below), although this is not usually recommended. It is common to deploy these devices in pairs for redundancy (because if the hub/switch fails, the entire service goes down).
- **Message Platform**: This is where email, voicemail or other messages are stored. On some systems voicemail and email are stored on separate servers
- **Directory Server**: This contains information on each subscriber needed to complete calls, send notifications, leave messages, etc. It may also contain the "rules" governing each subscriber. For example, there may be a rule that calls to a subscriber before 5pm will ring at the subscriber's office phone and mobile phone simultaneously, or that the office phone will ring first, followed by the mobile phone, followed by transferring the call to voice mail, each after a certain number of rings. The directory server contains the numbers and other information which the system needs about each subscriber to carry out the functions according to these rules.
- **Subscriber Database**: This contains detailed information on each subscriber, including their profile, account information, etc. On some systems the directory server and subscriber database may exist on the same physical hardware.
- **AAA Server**: This authenticates users when they log into the system, determines what permissions they have on the network (for example, which fields in their own profile they can change, how much disk space is allocated to them, etc.), and generates accounting/billing records which can be used by a separate billing system (not shown) to generate invoices
- **Web Server**: This contains the web site for the service, including each user's web interface for setting rules and preferences, viewing usage or billing information, accessing email or voicemail messages, etc.
- **Other Servers**: Depending on the system, other servers may be needed or may be added as an option. For example, some systems support speech recognition, and this function is assigned to a specialized server rather than integrated into the communications server (this is often done for performance reasons or to keep the cost of the basic system down). Also, it is quite common to have backup servers for each of the functions above in order to prevent data loss and service interruptions. Some systems support other functions such as information services (stock quotes, traffic reports, etc.) which require additional devices.
- **Network Management**: Each of the devices in the system is set up and monitored from a central workstation. The network management system allows the system administrator to set up the network, diagnose problems, upgrade software, recover from failures, etc.

- **Subscriber Management**: This is used to provision new users, enter or change subscriber information, access billing information, and perform other functions relating to individual subscribers. It is usually separate from the network management station for performance and security reasons.
- **Router**: This is a "gateway" to the Internet, and is used to receive or send email, allow access to web servers, etc. It may include a firewall or other security functions, or the firewall may be a separate device.

In addition to the unified messaging system itself, as described above, it would also be necessary to create and staff a call centre/customer service centre. This is a significant expense but is necessary in order to take orders, respond to customer enquiries, etc. The call centre would be somewhat integrated with the unified communications system (e.g. for accessing subscriber profiles, creating new users, accessing billing information, etc.).

Call Scenario

In order to understand how the system works as a whole, this section provides an example of a typical scenario for a "single number" service. In this example, the user has subscribed to the service and received a new phone number, which he has provided to all of his friends, family, business associates, etc. He has visited your website (which is located on the web server), entered his name and the password assigned to him when he subscribed, and has accessed his personal "presence management" page where he has set up the following rules:

1. Between 8am-6pm on weekdays, incoming calls will ring on his mobile phone and at the office. After 5 rings, calls will be transferred to his voicemail
2. At all other times, incoming calls will go directly to voicemail, unless the call is coming from any of the numbers he has designated as "friends", in which case the call will ring on his mobile phone and residential phone
3. All fax calls will go to his office fax machine
4. Any emails received which are marked "urgent" will cause the system to phone his mobile number with a recorded alert and allow him to have the email read to him using a text to speech function.
5. When any of the stocks he has listed in his profile drop in price by more than 10% from the previous day, the system will send an SMS message to his mobile phone

In addition to these "static" rules, he can also change the rules depending on his personal status. For example, when he goes into a meeting, he uses the web interface on his mobile phone to tell the system he is unavailable for the next hour. For any calls that come in during that hour, callers will receive a recorded message (in his voice) saying that he is unavailable and asking if they want to leave a message.

Most unified communications systems would be able to manage most or all of the situations and rules above, in addition to all kinds of other possibilities. Exact capabilities and limitations vary widely from system to system.

In this scenario, when someone dials the subscriber's number during business hours, the following things occur in the system:

1. The communications server receives the incoming call signal (i.e. the "ring") and determines who is being called and who the call is from (using called number and caller ID information supplied by the PSTN switch).
2. The communications server sends a request to the AAA server to confirm the number dialled belongs to an active subscriber.
3. The AAA server finds the number in its database and tells the communications server to answer the call.
4. The communications server "answers" the call, and determines if it is a voice or fax call.
5. If it is a fax, it stores the transmission in memory and sends a request to the directory server for this subscriber's fax number. It then dials the fax number and transmits the stored fax.
6. If it is a voice call, the communications server plays a short message to the caller (something like "please wait while your call is connected") and checks the subscriber's rules in the directory or subscriber database. The server then makes two outgoing calls to the numbers supplied by the directory server (one to the subscriber's business line and the other to his mobile phone).
7. If the subscriber answers one of the two lines, it connects the call and stops ringing at whatever location was not answered. If the subscriber does not answer after 5 rings, it drops both calls and connects the caller to its voicemail system located on the message server.
8. Once the call is complete, the system logs the time, duration and other call information which can be used for reports, billing, etc.

This is a simplified description of what happens in one call scenario. Different systems may handle calls in different ways, but this should provide a high level view of how the service actually occurs.

Typical Capital Costs

The costs for a unified communications system vary widely depending on the number of subscribers, features offered, etc. as well as the specific vendors chosen. Initial capital costs would likely be in the range of USD $500,000-$750,000, including all hardware, software, etc. A trial system (for example, supporting 1 PRI for a non-commercial service) would probably cost between $75,000-$150,000.

The majority of capital costs (approximately 70%-75%) are in hardware (the physical servers, workstations, network equipment, racks and cabling, power, cooling, etc.), with the remainder in software.

These estimates do not include other business costs such as facilities, power, telecommunications costs, administration, call centre, marketing, salary, etc. A start-up venture in this business would typically require total funding of USD $5-$10 million at a minimum (generally acquired through some combination of private investment, venture capital, loans, government incentives, etc.).

Conclusion

Unified communications is a high growth, high potential area which offers opportunities and possibilities for new communications services for existing and new providers and value-added resellers.

Understanding the technology is only the first step, but understanding these basics is important for anyone undertaking a more detailed market investigation and developing a business or investment plan.