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A Global Packet-Phone Network for Global Enterprises-Theory and Practice-

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-Theory and Practice-

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ABSTRACT

In the last decay, there has been many information and Internet technologies advances. Due to the advances, enterprises are now easier to be scattered throughout the world to closely reach their customers. However, the distributed global enterprises have to pay the high cost for telecommunications among the globally distributed sites.

In this paper, we propose a global packet-phone network that could be built efficiently with low cost for global enterprises to provide international telecommunications, fax, video conferences, and other value added services.

The proposed system has been experimented and used by many global enterprises. This paper describes the theory of the global packet-phone network, and discusses how the system is built in practice.

INTRODUCTION

In the last decay, there have been many advances in information technologies introduced, for example, fiber technology [6], FDDI LAN [7], ATM, Gigabit Ethernet, mobile and wireless communications [1], voice over FDDI/Ethernet [8], voice over IP [2], etc.

Among the technologies, Internet is an essential infrastructure for a global enterprise to build more effective operations and advantages. The development of Internet has been one of the most important activities [5] in the last decay. Internetresearch is evolving into the next generation Internet [3] and may provide real time applications capabilities.

As the global economy evolves, all enterprises face the competition from the whole world. Dealing with the global market has been the trend for all enterprises to adopt. Due to the advances in information technologies, enterprises are now easier to be scattered throughout the world to closely reach their customers. However, the distributed global enterprises have to pay the high cost for telecommunications among the globally distributed sites. Any enterprise which takes advantages of recent technologies advances can compete better in the global market [4].

The Evolution of telecommunication technologies have been from analog transmission, to digital transmission and then from circuit switching to packet switching.

The advantages of packet switching are of many folds: (1) better bandwidth utilization, (2) higher transmission rate, (3) capable of value-added services, (4) allows always-on network connections, …etc.

In this paper, we propose a global packet-phone network that could be built for each distributed global enterprise to cut down their international telephone cost and provide more value added services, such as fax and video conferences.

This paper is organized as follows. In the following session, we propose the components and configuration of the network, as well as the global addressing and dialing schemes. Then we propose the database design for packet-based virtual circuit management. Next, we propose the call setup, call in progress and call termination procedures for each packet-phone conversation.

The proposed global packet-phone network has been partially implemented and used by many global enterprises. We then describe the design of high performance packet-phone exchange device, and finally discuss the practice and implementation of the system.

THE GLOBAL PACKET- PHONE NETWORK

In this section, we describe the packet-based network for enterprise's world-wide telecommunications services.

The GPN Configuration

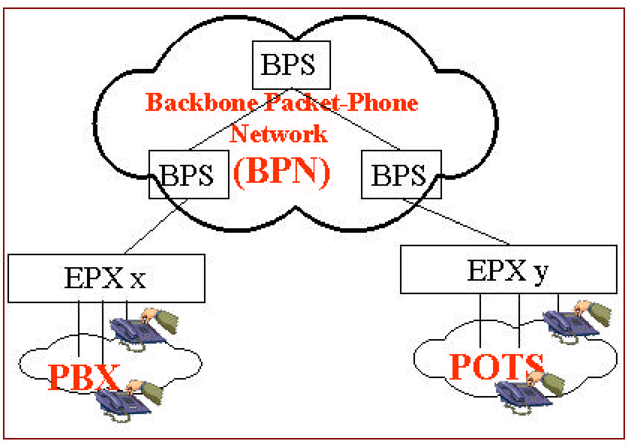

Figure 1: The Global Packet-Phone Network

The proposed Global Packet-Phone Network (GPN) consists of the following elements:

- Enterprise Packet-phone Exchange (EPX)
- Packet-Phone Set (Packet-Phone, for short)
- Enterprise Packet-phone Network (EPN)
- Backbone Packet-phone Switch (BPS)
- Backbone Packet-phone Network (BPN)

EPX and Packet-Phone Set

The most fundamental elements of our GPN are the enterprise packet-phone exchange, EPX, and packet-phone set. Each EPX provides voice call connection services upon user requests: (1) establishes connections for the calling party, (2) disconnects connections when conversations end, and (3) converts between voice signals and voice packets when conversations go on.

The EPX serves as the interface between our GPN network and the outside world. Therefore, each EPX has interfaces to be connected to our backbone network, and has packet-phone ports for voice calls to come in. The packet-phone set is the device that is attached to EPX exchange equipment and allows users to make packet-phone calls.

The packet-phone set is supposed to be able to perform voice packetizing task, but we propose the system in a way that has EPX to implement the voice signal and voice packet conversion. By this way we obtain better flexibilities of the choices of packet-phone sets. As a consequence, we make the whole system easier to implement with lower cost and much more packet-phone sets availability and external system interconnection choices.

In order to provide flexibility to interface our GPN with outside world, EPX is designed to have at least three types of interfaces for packet-phone sets to connect to EPX: (1) directly connect to EPX, (2) connect to EPX through Private Branch Exchange (PBX), or (3) connect to EPX through widely available telephone systems, e.g., the Plain Old Telephone Systems (POTS), as depicted in the figure.

EPN

Given an enterprise office site with EPX and packet-phone sets, there is a need of a local network to interconnect them as a whole. Within the enterprise site, the local network that interconnects the local EPX and all local packet-phone sets is called Enterprise Packet-phone Network (EPN).

For local wiring, we may adapt the CIA/TIA wiring standards for direct connection wiring purpose. We may integrate with the existing PBX or POTS wiring and systems to extend services to users. The configuration of EPN and the protocol standards to be deployed depend on the existing systems, say PBX, POTS etc., to be integrated.

With an EPN, then the enterprise is able to integrate packet switching technologies for local communications services. However, it is better off if we can extend the communications into global services.

BPS and BPN

To extend our packet-phone connection into global services, there is a need to develop a global backbone network in order to integrate the distributed enterprise sites over the world. The global backbone network should be able to interconnect with our EPX (enterprise packet-phone exchanges), and therefore, integrate them as a whole for global communications services.

The proposed global backbone network is called Backbone Packet-Phone Network (BPN). The simplest configuration of BPN is a collection of communications links that interconnect all EPX of a global enterprise. The BPN, in general, is a collection of Backbone Packet-Phone Switches (BPS) interconnected by communications links. The links are of free choice to the service providers. The communications links could be T1, T3, ATM, Gigabit Ethernet, …, etc. The Backbone Packet-phone Switches are also free of choice to the service providers, and can be IP routers, ATM switches, …, etc.

The BPN is designed as a transparent carrier network to deliver the packet-phone packets upon EPX requests. The standards of BPN are also free of choice to the service providers as long as the backbone network can interconnect with EPX exchanges and deliver the packet-phone packets transparently.

Both EPX and BPS have capabilities of performing switching functions. When we deal with switching functions, we may call *packet-phone switches* to refer to both BPS and EPX for convenience.

The Design of Global Addressing Scheme

In our packet-phone network, both EPX and BPS are of switching capabilities. For switching purpose, every packet-phone switch, either EPX or BPS, is assigned a globally addressable ID, say s. For convenience, each packet-phone switch can be denoted by [s].

To provide packet-phone service to users, each EPX has also exchanging capabilities and acts as the exchange interface between our packet-phone network and the outside world, say PBXs or POTS, mobile phones, etc. For this purpose, each EPX is designed to have voice connection ports, say p_1, p_2, …, p_n. Therefore, each voice connection port of an EPX, (hence, each packet-phone set connected to this port as well,) can be locally addressable by [p], where p is the port ID. This packet-phone also can be globally addressable by [p, s], where [s] is the global address of the locally attached EPX.

Given the globally addressable IDs, each packet-phone switch pair, say switch a and switch b, can then setup a communication channel in between, denoted by $[s_a]$-$[s_b]$, to provide packet-based communication services. For our packet-phone services purpose, the communication channel $[s_a]$-$[s_b]$ can be established by identifying a path between switches a and b for each voice call packet to be delivered, and by the switching functions of BPS along the path to perform (cut-through or store-and-forward) switching functions.

Symbolically, for each voice packet of a given voice call, there is a specific virtual path $[s_a]$-$[s_b]$ for delivering the voice packet. In fact, this specific virtual path corresponds to a switching path $[s_a]$-$[s_1]$-$[s_2]$-$[s_3]$-…-$[s_m]$-$[s_b]$, where $[s_1]$, $[s_2]$, $[s_3]$, …,$[s_m]$ are the backbone packet-phone switches along the path for the voice packet between switches a and b. Due to the advantages of packet switching, all packets of a given phone call may be switched via different routes to provide better routing flexibility.

The Design of Global Dialing Scheme

In this section, we describe the global dialing scheme for each packet-phone to make phone calls. As mentioned earlier, each packet-phone in our network can be uniquely identified by a pair of IDs, [p,s], where [s] is global address of the enterprise packet-phone exchange that the packet-phone is attached to, and [p] is the associated port ID on the enterprise packet-phone exchange.

For packet-phone x=$[p_x, s_x]$ to call packet-phone y =$[p_y, s_y]$, in our packer-phone network, packet-phone x simply dials the destination address $[p_y, s_y]$ of packet-phone y. Then our backbone packet-phone network BPN will automatically establish paths between exchanges s_x and s_y for communications between the two packet-phones.

Upon receiving the call request, packet-phone y also receives the global address of the caller, packet-phone x. Hence, packet-phone y can also call back to packet-phone x. It is worthwhile to mention that the virtual channel from x to y is not necessary to be the same as that from y to x. Furthermore, each voice packet during the conversation connection is not necessary follow the same route in our backbone network because we may utilize the advantages of packet switching technology to provide flexible routes.

DATABASE AND PACKET-PHONE VIRTUAL CIRCUIT MANAGEMENT

Distributed Virtual Circuit Management Database

For packet-phone service purpose, we designed a *distributed virtual circuit management database* in the system. With this database design, for each pair of packet-phone conversation, we have each corresponding enterprise packet-phone exchange EPX to maintain a database that stores one entry, called the *packet-phone virtual circuit, conversation virtual circuit, or virtual circuit (VC),* for short. Each VC is stamped with a *serial number* for control. This virtual circuit entry is added into the database once the conversation is activated, and is removed if the conversation ends.

Note that we do not need a dedicated physical circuit for each of our packet-phone conversations. The virtual circuit of a conversation ends if a termination signal is received (including a hang up signal from any calling party, or a termination control packet from the remote end), or if no voice packets and no control packets received from the remote end during conversation time out interval T, to be described in the following session.

The Schemes for Silent Mode Management and Failure Recovery

The difference between our packet-phone and the traditional one is that the connection between two packet-phones does not really have a dedicated physical circuit, instead, only a *virtual circuit* for each voice packet transmission. Serious problems then arise: *(1)* How do we distinguish a virtual circuit that has no packet traffic is really terminated or just because that call parties do not speak at all? *(2)* Will a packet-phone and the connected port be hanged on because of some unexpected or process failure problems?

The answer to the first question is as follows. If the caller is still on the line and the caller just keeps silent, then the associated EPX is designed to send out *silent packets* during the caller's *silent mode*. Then even the conversation parties are in the silent mode, we have silent packets to tell us that the conversation is still on going. Hence, we can ensure that the packet-phone virtual circuit will be kept alive during the silent mode.

To answer the second question, we set a *conversation time out timer*, say *T*, in the system. Each EPX is asked to do the following: For each packet-phone virtual circuit, the EPX must either send out enough, say *k*, voice packets or silent packets within each time interval T. Then each malfunction virtual circuit will be successfully terminated by time out timer even if the termination packets are not received for any reason.

PACKET-PHONE CALL MANAGEMENT

The packet-phone must simulate the way people place the traditional phone calls. As traditionally, there are three basic procedures to be handled for each call, namely, call setup, call in progress and call termination. Besides, we propose a methodology to improve the virtual circuit quality to meet users' expectations.

Call Setup

Each packet-phone call is initiated by call setup. In our packet-phone network, each packet-phone set can be locally identified by [p], where p is the port ID of the connected EPX, and can be globally identified by [p, s], where [s] is the globally addressable address assigned to the EPX.

The following table shows the procedures that caller x places a call to callee y.

Table 1: The Call Setup Procedure

	Caller x	EPXx	BPN (Backbone Packet-Phone Network)	EPXy	Callee y
x picks up	....	→*			
	Dial tone	←*			
	Dial [p_y, s_y]	→*			
	Call tone	←*→	Call setup (n, [p_x,s_x], [p_y,s_y])	→*	
Ring	Ring tone	←*←	Ring (n,[p_y, s_y], [p_x, s_x])	←*→	Ring tone
Line Busy	Busy tone	←*←	Busy (n,[p_y, s_y], [p_x, s_x])	←*→	*INT* tone
y picks up				*←	....
	Call tone	←*←	Call tone (n,[p_y, s_y], [p_x, s_x])	←*→	Call tone
		VC		VC	

In the table, caller x places a call to callee y. The Enterprise packet-phone exchange (EPX x) and the packet-phone port that caller x is attached are [s_x] and [p_x], respectively. Therefore, the packet-phone of caller x can be globally addressed by packet-phone address x=[p_x, s_x]. With the same argument, the global address of callee y's packet-phone is y=[p_y, s_y], where [s_y] and [p_y] are the address of the corresponding enterprise packet-phone exchange EPX y=[s_y] and port ID, respectively.

Step1: Caller x picks up the phone set

(1-1) At the beginning, caller x picks up the packet-phone set and sends a pick-up signal to EPX x = [s_x].

(1-2) Upon receiving the phone pick-up signal, EPX x generates dial tones to caller x, and then waits for the caller to dial the destination number.

(1-3) Caller x dials the destination address [p_y, s_y] to EPX x.

Step 2: Virtual circuit between EPX x and EPX y

(2-1): We are now ready to establish a virtual circuit between the local exchange EPX x and the destination exchange EPX y.

Given the dialed destination number [p_y, s_y] from caller x, EPX x generates a serial number, n, to establish a new virtual circuit between EPX x and EPX y. Then EPX x generates two types of packets as follows. First, EPX x generates a series of call tones to caller x to ask caller x to wait for the response from the other end. Second, EPX x generates the call setup packets, "Call setup (n, [p_x, s_x], [p_y, s_y])", and transmits to the destination enterprise packet-phone exchange EPX y=[s_y], via Backbone Packet-phone Network (BPN).

(2-2) **Case a: The line of Callee y is available**

Then exchange EPX y generates ring tone signals and informs callee y that an incoming call arrives. EPX y also generates ring tone packets to EPX x via the backbone network BPN. Upon receiving ring tone packets from EPX y, exchange EPX x also generates ring tone signals to caller x, and tells caller x to wait for the remote party to pick up the phone set.

Case b: The line of Callee y is busy

In this case, exchange EPX y sends a interrupt signal to callee y to inform that there is an incoming call interupt. Furthermore, exchange EPX y generates busy tone packets and transmits via BPN to exchange EPX x. Upon receiving the busy tones, EPX x generates busy tone signals to inform caller x that the other end is busy.

Step 3: Callee y picks up the packet-phone set

(3-1) When callee y picks up the packet-phone set, there will be a corresponding pick-up signal sent to EPX y.

(3-2) When the phone pick-up signals are received, exchange EPX y replies to callee y with call tone signals. At the same time, EPX y generates call tone packets to EPX x. And, EPX x then generates call tone signals to caller x to inform that the remote end picks up the packet-phone already.

(3-3) After both ends of the call party are informed with call tone signals, then each EPX adds a corresponding virtual circuit (VC) entry into its VC management database.

Call Termination

Recall that our packet-phone network does not need to reserve a dedicated physical circuit for each packet-phone service. Hence, during the conversations, we maintain the virtual circuit by monitoring not only voice packets generated by the conversation, but also the silent packets during the silent mode of the conversation. Therefore, besides the callers to hang up the packet-phones, we also need to terminate the virtual circuit if the network fails, or the conversation operations malfunction and do not follow the protocol definitions.

The call termination procedure can be divided into the following cases:

Case 1: Call Hang up:
We state the procedure for the case that caller x hangs up the

phone set. It is similar for the case of callee y.

(a) Local Site

- EPX x terminates the connection between itself and the phone set of caller x, and removes the corresponding conversation virtual circuit entry (n, [p_x, s_x], [p_y, s_y]) from the VC management database.
- EPX x generates termination packets "terminate (n, [p_x, s_x], [p_y, s_y])" to the remote exchange EPX y via BPN.

(b) Remote Site

- Upon receiving the termination packets, EPX y disconnects the connection between itself and the phone set y , and removes conversation virtual circuit (n, [p_y,s_y], [p_x,s_x]) from the VC management database.

Case 2: Conversation Timer *T* Times out

For any conversation virtual circuit still maintained in the VC management database, the enterprise packet-phone exchange must ensure that the virtual circuit is actually alive. If no voice traffic or silent packets received d uring the conversation timer T, then we terminate the conversation virtual circuit by:

- Remove the virtual circuit entry from the database.
- Disconnect the connection between the enterprise packet-phone exchange and the corresponding phone set.
- Generate call termination packets and send to the remote site EPX via BPN so that the other end may terminate the *corresponding* virtual circuit ASAP.

Call Quality Improvement with Silent Signals

In the packet-phone conversation process, we usually have just one end of the two is talking, and the other is only listening. Note that no voice signals at all come out from the silent side. This makes the talking side feel that the link seems disconnected.

To overcome this problem, we suggest that, if any end is in silent mode, then the corresponding EPX must *frequently generate silent packets* to the talking side EXP. The talking side EXP then generate silent signals to the talking side phone set. We also suggest that the silent signals are converted to *audible "sound of the environmental background*" so that the talker can hear and feel that the other end is with him or her.

Call in Progress

During the call in progress phase, both parties of call may speak as wish.

Case 1: Caller x Is Speaking (The same for Callee y)
In case caller x is talking, then enterprise packet-phone exchange EPX x must encode the voice signals into voice packet format, and transmit the converted voice packets to the destination packet-phone exchange EPX y via BPN. Upon receiving the voice packets, EPX y then decodes the voice packets back to voice signals to callee y. So, callee y hears what caller x speaks.

Case 2: Caller x Is Silent (The same for Callee y)
That is the case that any party are kept silent for a long enough time. In this case, enterprise packet-phone exchange EPX x generates silent packets to the remote end*from time to time*, so that the remote EPX y is sure that caller x is still on the line. Upon receiving the silent packets, EPX y then sends silent signals to callee y s o that callee y knows that the other end is listening.

Table 2: Call in Progress Procedure

	Caller x	**EPXx**	**BPN (Backbone Packet-phone Network)**	**EPXy**	Callee y
From caller	Voice	→*→	Voice (n, [p_x, s_x], [p_y, s_y])	→*→	Voice
	Silent	→*→	Silent (n, [p_x, s_x], [p_y, s_y])	→*→	Silent Tone
From callee	Voice	←*←	Voice (n, [p_y, s_y], [p_x, s_x])	←*←	Voice
	Silent Tone	←*←	Silent (n, [p_x, s_x], [p_y, s_y])	←*←	Silent

THE DESIGN OF HIGH PERFORMANCE EPX

The most critical element of our GPN is EPX, the exchange device located at each site of enterprises. Usually, we may use a computer, or even just a PC, to act as an EPX by adding voice cards to provide phone set port connections. The use of PC with voice cards provide an experimental device for voice digitization. However, the PC model always generates high heat, causes unstable system operations and is easy to fail and difficult to recover.

In our practice, the EPX is a diskless system with Ethernet ports and voice connection ports. The Ethernet ports allow the device to easily integrate with Internet protocols. Each of the voice port has a dedicated high performance signal processing chip for high speed conversion between voice signals and packets. With this high performance chip and system design, each EPX always provides high performance as well as reliable system operations.

The EPX allows system administrators to setup *dial plans* for each global enterprise, based upon the global addressing and dialing schemes described in previous sessions. In other words, we first assign a globally addressable ID to each of the enterprise EPX, assign a dial number to each packet-phone set, and then we can setup a dial plan for the enterprise accordingly.

In the implementation, each EPX ID is composed of a *network address* and an *exchange number*. The network address is used to connect the EPX to the backbone network, and only the system administrators need know the network address of the EPX. The exchange number is the number for users to dial and make phone calls. In other words, an EPX is simply a telecommunications exchange device from users' viewpoint.

Each packet-phone set is attached to an EPX voice port. Each EPX voice port is also assigned an ID. This ID is used as the ID assigned to the connected packet-phone set. Therefore, each packet-phone set is addressable by the pair of exchange number, x, and port ID, p. Therefore, [s, p] is the global address of the packet-phone set, and is the number for users to dial and make phone calls in practice.

As mentioned earlier, EPX has to take care of the silent mode situation. The silent side EPX sends out silent packets to the other end so that the talking party can feel that the other end is listening, and is with him or her.

IMPLEMENTATION PRACTICE OF GLOBAL PACKET-PHONE NETWORK FOR GLOBAL ENTERPRISES

A global enterprise usually deals with the whole world as the target market. As a consequence, a global enterprise usually has distributed office sites all over the world, and has a big amount of *international telecommunications cost*. To compete in the global market, an aggressive global enterprise usually: *(1)* has each of the distributed office sites be connected to *Internet*, or *(2)* has *leased lines* for intra-enterprise telecommunications among the distributed office sites.

In this session, we point out how a global enterprise can be benefited from using the packet switching technology for global packet-phone services. Given the description in the above sessions, we describe how to build a GPN for global enterprises that is effective, low cost and could be fast implemented.

GPN for Global Enterprises over Internet

Due to information advances, Internet connection becomes an essential infrastructure for global business operations. More and more global enterprises are now connected to Internet, and hence satisfy the case 1 described above.

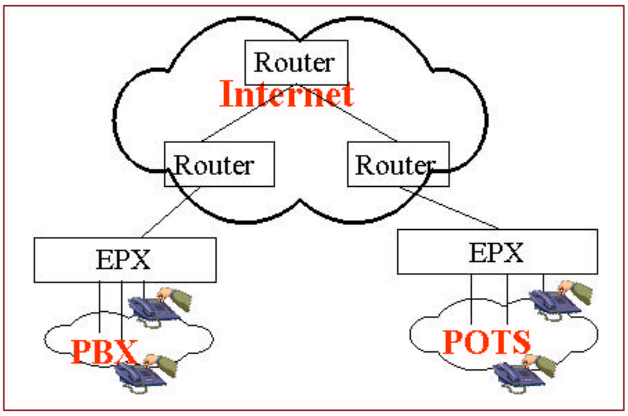

Figure 2. GPN over Internet

For each enterprise site, there is a local area network for all users to get onto Internet. In our experiment, the most critical place that communications congestion arises is the local area network of each local site. Most local networks implement Ethernet networking, and therefore, cause collisions and long communications delay if the traffic load is heavy.

To deal with the local area network traffic congestion problem, we suggest introducing a traffic manager between the local area network and the Internet backbone. In other words, we suggest giving higher priority and more bandwidth to voice packets than the other services.

Internet is a global resource for any body to use, and transmits packets across Internet with best effort. The above GPN over Internet Figure provides an ideal configuration for such global enterprise to start with. Given the Internet connections, we may use Internet as the backbone packet-phone network, and use IP routers as the backbone packet-phone switches. The advantages of this configuration is the low cost, fast implementation and global reach through Internet.

The disadvantages of this configuration are: Internet does not guarantee the delivery of packets to the destination within a specified time interval, and does not guarantee at all whether the packets will be eventually delivered to the destination. That is, these disadvantages mean that Internet does not guarantee to meet the requirements for telecommunications conversations. In practice, however, one can still use Internet as a possible enterprise packet-phone backbone.

In implementation, Internet is a collection of IP routers and telecommunications links. Each telecommunications link is serial transmission, and therefore, has first-in-first-out property. Each backbone IP router is usually a high performance device with queuing capability. Therefore, once voice packets get on Internet backbone, it is very likely that they may follow first-in-first-out property and arrive at the destination end with a time delay.

In practice with above mentioned EPX designs, if the time delay between two sites of a global enterprise is less than 200 ms, then the GPN over Internet solution provides good enough quality for voice communications services. And the resulting GPN provides great packet-phone service quality when traffic manager is appropriately introduced as described above. Furthermore, the proposed GPN can provide not only telephone services, but also fax communications. When work with Internet and video transmission, GPN may also provide video conference solutions and many other value added services.

GPN for Global Enterprises with Leased Lines for Intra-telecommunications

If the a global enterprise has leased lines for intra-enterprise telecommunications already, then it is rather easy to implement our GPN to provide packet-phone services for the global enterprise.

The easiest way to construct a GPN for such enterprise is using these leased lines to build a *private Packet-Phone Network Backbone*. By this way, we may easily maintain the packet-phone quality of the private BPN by managing network utilization and transmission delay. Therefore, we may guarantee the delivery and the delay time of all voice packets to meet the telecommunications requirements. In other words, the resulting GPN provides excellent quality packet-phone global services with minimum added cost. Furthermore, this private BPN can also serve as the private internet and connect to the global Internet for more value added services.

CONCLUDING REMARKS

In this paper, we propose a Global Packet-phone Network (GPN) to provide global telecommunications services. This GPN network service utilizes more efficient connectionless packet-phone technologies, and may be built for global enterprises to provide global telecommunications services and enhance the competitive advantages.

Each packet-phone conversation does not need to reserve a dedicated circuit. Hence, it is more efficient for resource utilization, and needs different ways to manage each packet-phone conversation. The proposed GPN has a conversation virtual circuit database to manage each packet-phone conversation. For each packet-phone conversation, we describe the essential call handling procedures, namely, call setup, call in progress and call termination to handle packet-based, connectionless conversation virtual circuits.

The connectionless property of packet-phone provides better resource utilization. However, in *silent mode*, there will be no voice packets generated at all, and cause the virtual circuit to be terminated improperly. A special scheme introduced in our procedure is the *silent packets* to provide a "*background noise*" to the talker so that the talker knows that the other end is really with him or her and listening.

We describe, for practical implementation, how to utilize Internet technologies and existing Internet resources to build GPN for global enterprises. In practice, we may use existing intra-enterprise telecommunications links or use existing Internet access as the packet-phone backbone for low cost and speedy implementation.

Our experiment shows that if the network bandwidth is properly managed for the voice packets in each local area network, and the ping delay between two sites of the global enterprise is less than 200 ms, then the proposed GPN provides excellent packet-phone services quality.

Furthermore, this GPN can provide not only telephone services, but also fax communications. When work with Internet and video transmission, then GPN may also provide video conference solutions, and many other value added services.

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