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Chapter 1 : Cisco Voice over Frame Relay, ATM, and IP

*Optimizing Voice in ATM/IP Mobile Networks [Juliet Bates] on calendrierdelascience.com *FREE* shipping on qualifying offers. As telecom engineers struggle with implementing 3G networks, this in-depth guide to advanced methods of transmitting voice and modulated data over 3G ATM/IP backbones is especially timely.*

However, the use and exact values of the GFC field have not been standardized, and the field is always set to If 0, user data cell and the following apply: Used by AAL5 to indicate packet boundaries. If the most significant bit MSB of the PT field is 0, this is a user data cell, and the other two bits are used to indicate network congestion and as a general purpose header bit available for ATM adaptation layers. If the MSB is 1, this is a management cell, and the other two bits indicate the type. Network management segment, network management end-to-end, resource management, and reserved for future use. The 8-bit CRC is used to correct single-bit header errors and detect multi-bit header errors. When multi-bit header errors are detected, the current and subsequent cells are dropped until a cell with no header errors is found. All four GFC bits must be zero by default. Synchronization is also maintained at AAL1. Which AAL is in use for a given cell is not encoded in the cell. Instead, it is negotiated by or configured at the endpoints on a per-virtual-connection basis. Following the initial design of ATM, networks have become much faster. A byte bit full-size Ethernet frame takes only 1. Some consider that this makes a case for replacing ATM with Ethernet in the network backbone. However, it should be noted that the increased link speeds by themselves do not alleviate jitter due to queuing. Additionally, the hardware for implementing the service adaptation for IP packets is expensive at very high speeds. Reasons for virtual circuits[edit] ATM operates as a channel-based transport layer, using virtual circuits VCs. This is encompassed in the concept of the virtual paths VP and virtual channels. The length of the VPI varies according to whether the cell is sent on the user-network interface on the edge of the network , or if it is sent on the network-network interface inside the network. The function of the VCI is similar to that of the data link connection identifier DLCI in frame relay and the logical channel number and logical channel group number in X. The VPI is useful for reducing the switching table of some virtual circuits which have common paths. When an ATM circuit is set up each switch on the circuit is informed of the traffic class of the connection. ATM traffic contracts form part of the mechanism by which " quality of service " QoS is ensured. There are four basic types and several variants which each have a set of parameters describing the connection. CBR - Constant bit rate: VBR - Variable bit rate: ABR - Available bit rate: UBR - Unspecified bit rate: VBR has real-time and non-real-time variants, and serves for "bursty" traffic. Non-real-time is sometimes abbreviated to vbr-nrt. Most traffic classes also introduce the concept of Cell-delay variation tolerance CDVT , which defines the "clumping" of cells in time. Traffic policing[edit] To maintain network performance, networks may apply traffic policing to virtual circuits to limit them to their traffic contracts at the entry points to the network, i. If the traffic on a virtual circuit is exceeding its traffic contract, as determined by the GCRA, the network can either drop the cells or mark the Cell Loss Priority CLP bit to identify a cell as potentially redundant. Basic policing works on a cell by cell basis, but this is sub-optimal for encapsulated packet traffic as discarding a single cell will invalidate the whole packet. As a result, schemes such as partial packet discard PPD and early packet discard EPD have been created that will discard a whole series of cells until the next packet starts. This reduces the number of useless cells in the network, saving bandwidth for full packets. Traffic shaping[edit] Traffic shaping usually takes place in the network interface card NIC in user equipment, and attempts to ensure that the cell flow on a VC will meet its traffic contract, i. Since the reference model given for traffic policing in the network is the GCRA, this algorithm is normally used for shaping as well, and single and dual leaky bucket implementations may be used as appropriate. Types of virtual circuits and paths[edit] ATM can build virtual circuits and virtual paths either statically or dynamically. Static circuits permanent virtual circuits or PVCs or paths permanent virtual paths or PVPs require that the circuit is composed of a series of segments, one for each pair of interfaces through which it passes. They also do not support the re-routing of service in the event of a failure. ATM networks create and remove switched virtual circuits SVCs on demand when requested by an end piece of equipment. One application for SVCs is to carry individual telephone calls when a network of telephone switches are inter-connected using ATM. PNNI also includes a very powerful summarization mechanism to allow construction of very large networks, as well as a call admission control CAC algorithm which determines the availability of sufficient bandwidth on a proposed route through a network in order to satisfy the service requirements of a VC or VP. Call admission and connection establishment[edit] A network must establish a connection before two parties can send cells to each other. It can be a permanent virtual circuit PVC , which is created administratively on the end points, or a switched virtual circuit SVC , which is created as needed by the communicating parties. SVC creation is managed by signaling , in which the requesting party indicates the address of the receiving party, the type of service requested, and whatever traffic parameters may be applicable to the selected service.

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Chapter 2 : ABIS OPTIMIZATION â€“ Emtel Communications

As telecom engineers struggle with implementing 3G networks, this in-depth guide to advanced methods of transmitting voice and modulated data over 3G ATM/IP backbones is especially timely. Readers will find it loaded with valuable data and real world insight based on actual implementation experience.

Show Context Citation Context Summary of the Pros and Cons of Hierarchical Architectures hierarchy. A hybrid scheme utilizing both hierarchical entries and pre-assigned home location registers HLRs is also possible. Assume that database entries are maintained only at selective nodes of the hiera Mobility management in current and future communication networks by Ian F. The integration of these networks is discussed in the context of the next evolutionary step of wireless communications networks. First, a review is The latest protocol changes for location registration and handoff are investigated for Mobile IP, followed by a discussion of proposed protocols for wireless ATM and satellite networks. Finally, an outline of open problems to be addressed by the next generation of wireless neiwork service is discussed. Current voice, fax, e-mail, and paging services will give way to data transfer, videoconferencing, image transfer, and video delivery, Show Context Citation Context The status notification procedures of the PNNI network protocol are exploited in order to propagate location information about each MT in the network without the use of a database. In this paper, we propose an algorithm for optimizing the route of a connection that becomes suboptimal due to operations such as handoffs and location-based reroutes, for mobile ATM asynchronous transfer mode networks based on the PNNI private network-to-network interface standard. We then apply this algorithm to the mobile location management problem. A comparative performance analysis of the oneand two-phase connection setup schemes is presented. Measures of comparison are call setup delay and the amount of network resources allocated to a connection. The maximum call setup delay worst case call setup delay is lower in the two-phase scheme, but the average call setup delay is lower in the one-phase scheme. The response indicates the exact location of the mobile, which allows for a direct connection to be set up on the optimal path. While this approach has the advantage of resulting in optimal connection paths which implies that a route optimization procedure is not required subsequently , it has certain disadvantages. Chang - in PCS. Abstract â€” Location management is important to effectively keep track of mobile terminals with reduced signal flows and database queries. Even though dynamic location management strategies are known to show good performance, we in this paper consider the static location management strategy which is Even though dynamic location management strategies are known to show good performance, we in this paper consider the static location management strategy which is easy to implement. A system with single home location register and pointer forwarding is assumed. A mobile terminal is assumed to have memory to store the IDs of visitor location registers VLRs each of which has the forwarding pointer to identify its current location. To obtain the registration point which minimizes the database access and signaling cost from the current time to the time of power-off probabilistic dynamic programming formulation is presented. A Selective Pointer Forwarding scheme is proposed which is based on one-step dynamic programming. The proposed location update scheme determines the least cost temporary VLR which point forwards the latest location of the mobile. The computational results show that the proposed scheme outperforms IS, pure Pointer Forwarding, and One-step Pointer Forwarding at the expense of small storage and a few computations at the mobile terminals. Hierarchical structure consists of a number of databases each of which is connected to others only throu

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Chapter 3 : USB1 - Voice over IP optimization for mobile IP - Google Patents

Optimizing Voice in ATM/IP Mobile Networks: Capacity and Quality (McGraw-Hill Telecom. Engineering) - Kindle edition by Juliet Bates. Download it once and read it on your Kindle device, PC, phones or tablets.

Standards such as H. The general maturity of standards has in turn generated robust protocol stacks that can be purchased "off the shelf" by vendors, further ensuring interoperability. Technology Recent advances in technology have also enabled voice integration with data. For example, new Digital Signal Processor DSP technology has allowed analog signals to be processed in the digital domain, which was difficult or impossible only a few years earlier. These powerful new chips offer tremendous processing speeds, allowing voice to be sampled, digitized, and compressed in real time. Further breakthroughs in the technology allow as many as four voice conversations to be managed at the same time on a single chip, with even greater performance in development. These technologies greatly reduce the cost and complexity of developing products and deploying voice over data solutions. Previously, it was assumed that voice quality would suffer as bandwidth was decreased in a relatively linear fashion. However, new, sophisticated algorithms employed in new codecs have changed that view. It is now possible to obtain reasonably good-sounding voice at a fraction of the bandwidth once required. More importantly, these new algorithms have been incorporated into the standards to allow interoperability of highly compressed voice. Network Performance Finally, data-networking technology has improved to the point that voice can be carried reliably. Over the last few years, growth in voice traffic has been relatively small, while data traffic has grown exponentially. The result is that data traffic is now greater than voice traffic in many networks. In addition, the relative importance of data traffic has grown, as businesses and organizations come to base more business practices and policies on the ubiquity of data networks. This increase in importance of data networks has forced a fundamental change in the way data networks are engineered, built, and managed. Typical "best-effort" data modeling has given way to advanced policy-based networking with managed quality of service to support an even greater range of applications. Voice traffic, as an application on a data network, has benefited greatly from these technologies. For example, support of delay-sensitive SNA traffic over IP networks resulted in breakthroughs in latency management and queuing prioritization, which was then applied to voice traffic. As stated previously, deployment of new technologies and applications must also be driven by greater demand from users. For example, digital audio tape DAT technologies never enjoyed widespread use outside the audiophile community because of the high cost and only marginally better perceived performance than analog tapes. Economic Advantages It has been estimated that packet voice networking costs only 20 to 30 percent of an equivalent circuit-based voice network. This is true for both carriers service providers and enterprise private users. The resulting savings in long-distance toll charges often provide payback in as little as six months especially if international calls are avoided. Using data systems to carry voice as "virtual tie lines" between switches is also useful to service providers. In fact, many new carriers have started to embrace packet-based voice technologies as their primary network infrastructure strategy going forward. It is also possible to switch voice calls in the data domain more economically than traditional circuit-based voice switches. For large, multisite enterprises, the savings result from using the data network to act as a "tandem switch" to route voice calls between PBXs on a call-by-call basis. The resulting voice network structure is simpler to administer and uses a robust, nonblocking switching fabric made up of data systems at its core. However, there are added benefits, which will become more evident in the future. As applications evolve, organizations will gain increased user productivity from the integration of voice and computer applications. Computer telephony integration CTI was begun by PBX vendors in the s to integrate computers with PBXs to provide applications such as advanced call center features for example, "screen pops" for agents. For example, Unified Messaging systems are now available that combine voice mail, e-mail, and fax messaging into a single, convenient system. With these advanced systems, users can have e-mail read to them over the phone or can add document attachments to voice mail. At the enterprise level, new applications such as virtual call centers allow call center agents to be distributed anywhere within reach of the data network, while still receiving the full suite of call center functions and features. They can even receive calls over their computers rather than using a traditional telephone instrument, and they can provide "blended contact center" support to answer Web user questions with electronic chat capability and e-mail between voice calls. These capabilities go far beyond simple cost savings and will ultimately make organizations much more effective and profitable. The strong pressures driving the integration of voice and data networks have resulted in various solutions to the problem, each with its own strengths and weaknesses. Three general approaches exist: These are illustrated in the below figure. More details are available later in this article. ATM offers many advantages for transport and switching of voice. First, quality of service QoS guarantees can be specified by service provisioning or on a per-call basis. Administration is similar to circuit-based voice networks. However, VoATM suffers from the burden of additional complexity and incomplete support and interoperability among vendors. It also tends to be more expensive because it is oriented toward all optical networks. Nevertheless, ATM is quite effective for providing trunking and tandem switching services between existing voice switches and PBXs. It benefits from much simpler administration and relatively lower cost than VoATM, especially when deployed over a private WAN network. When deployed over a carefully engineered Frame Relay network, VoFR works very well and provides good quality. However, voice quality over Frame Relay can suffer depending on network latency and jitter. Although minimal bandwidth and burstiness are routinely contracted, latency and jitter are often not included in service level agreements SLAs with service providers. As a result, voice performance can vary. For this reason, many large enterprise customers are beginning to specify latency and jitter, as well as overall packet throughput from carriers. In these situations, voice over Frame Relay can provide excellent service. Of all the packet voice technologies, VoIP has perhaps the most difficult time supporting voice quality because QoS cannot be guaranteed. Normal applications such as TCP running on IP are insensitive to latency but must retransmit lost packets due to collisions or congestion. Voice is much more sensitive to packet delay than packet loss. In addition to normal traffic congestion, QoS for VoIP is often dependent on lower layers that are ignorant of the voice traffic mingled with the data traffic. Voice Networking Basic voice technology has been available for more than years. During that time, the technology has matured to the point at which it has become ubiquitous and largely invisible to most users. Traditional analog telephone instruments used for plain old telephone service POTS use a simple two-wire interface to the network. This economical approach has been effective but requires special engineering regarding echo. Basic Telephony Three types of signaling are required for traditional telephony: Supervision monitors the state of the instrument-for example, allowing the central office or PBX to know when the receiver has been picked up to make a call, or when a call is terminated. Alerting concerns the notification of a user that a call is present ringing or simple call progress tones during a call such as busy, ringback, and so on. Finally, addressing enables the user to dial a specific extension. In addition to signaling, telephony services also provide secure media transport for the voice itself, analog-to-digital conversion, bonding and grounding for safety, power, and a variety of other functions when needed. Analog voice interfaces have evolved over the years to provide for these basic functions while addressing specific applications. The user side telephone expects to receive power from the network as well as supervision. A foreign exchange service FXS interface is used to connect an analog telephone, fax machine, modem, or any other device that would be connected to a phone line. It outputs 48 vdc power, ringing, and so on, and it accepts dialed digits. It is used to connect to a switching system providing services and supervision, and it expects the switch to provide supervision and other elements. The terms FXS and FXO were originally used within telephone company networks to describe provision of telephone service from a central office other than normally assigned. Typical telephones operate in a loop start mode. The telephone normally presents a high impedance between the two wires. When the receiver goes off-hook, a low-impedance closed circuit is created between the two wires. The switch, sensing current flow, then knows that the receiver is off-hook and applies a dial tone. The switch also checks to be sure that the receiver is on-hook before sending a ringing signal. This system works well for simple telephones, but it can cause problems on trunks between PBXs and COs with high activity. In that situation, the remote end and the CO switch can both try to seize the line at the same time. This situation, called glare, can freeze the trunk until one side releases it. The solution is to short tip or ring to ground as a signal for line seizure rather than looping it. This is called ground start. After the line is seized, it is necessary to dial the number. Normal human fingers cannot outrun the dial receivers in a modern switch, but digits dialed by a PBX can. In that case, many analog trunks use a delay start or wink start method to notify the calling device when the switch is ready to accept digits. This is a four- or six-wire interface that includes separate wires for supervision in addition to the voice pair. Analog voice works well for basic trunk connections between switches or PBXs, but it is uneconomical when the number of connections exceeds six to eight circuits. At that point, it is usually more efficient to use digital trunks. In North America, the T1 1. In other parts of the world, E1 2. Engineers refer to the adoption of E1 and T1 internationally as "the baseball rule"-there is a strong correlation of countries that play baseball to the use of T1. The first step in conversion to digital is sampling. The Nyquist theorem states that the sampling frequency should be twice the rate of the highest desired frequency. Early telephony engineers decided that a range of hertz would be sufficient to capture human voices which matches the performance of long analog loops. Therefore, voice channels are sampled at a rate of times per second, or once every ms. Each of these samples consists of an 8-bit measurement, for a total of bits per second to be transmitted. As a final step, companding is used to provide greater accuracy of low-amplitude components. In North America, this is u-law mu-law , while elsewhere it is typically A-law.

Chapter 4 : Asynchronous transfer mode - Wikipedia

A guide to advanced methods of transmitting voice and modulated data over 3G ATM/IP backbones, this work contains data and insight based on implementation experience.

This is accomplished, in part, by sending an agent advertisement identifying an H. A packet is received from the node, where the packet is addressed to the H. A packet including the requested IP address is forwarded to the node. An IP packet addressed to the IP address and including voice information may then be received from the node. A node visiting the Foreign is capable of sending IP packets including voice information and receives an agent advertisement identifying an H. The node sends a packet addressed to the H. A packet including the requested IP address is received by the node. The node may then send an IP packet addressed to the IP address and including voice information from the node. More particularly, the present invention relates to optimizing voice over IP in a Mobile IP environment. Without Mobile IP or related protocol, a Mobile Node would be unable to stay connected while roaming through various sub-networks. This is because the IP address required for any node to communicate over the internet is location specific. Each IP address has a field that specifies the particular sub-network on which the node resides. If a user desires to take a computer which is normally attached to one node and roam with it so that it passes through different sub-networks, it cannot use its home base IP address. As a result, a business person traveling across the country cannot merely roam with his or her computer across geographically disparate network segments or wireless nodes while remaining connected over the internet. This is not an acceptable state-of-affairs in the age of portable computational devices. To address this problem, the Mobile IP protocol has been developed and implemented. Both of these references are incorporated herein by reference in their entireties and for all purposes. A particular Mobile Node e. When the Mobile Node roams, it communicates via the internet through an available Foreign Agent. Presumably, there are many Foreign Agents available at geographically disparate locations to allow wide spread internet connection via the Mobile IP protocol. Note that it is also possible for the Mobile Node to register directly with its Home Agent. As shown in FIG. Note that Home Agent 8 need not directly connect to the internet. For example, as shown in FIG. Router R1 may, in turn, connect one or more other routers e. Now, suppose that Mobile Node 6 is removed from its home base network segment 12 and roams to a remote network segment Network segment 14 may include various other nodes such as a PC The nodes on network segment 14 communicate with the internet through a router which doubles as Foreign Agent Mobile Node 6 may identify Foreign Agent 10 through various solicitations and advertisements which form part of the Mobile IP protocol. For example, the attachment may be limited to a period of time, such as two hours. Now, suppose that Mobile Node 6 wishes to send a message to a corresponding node 18 from its new location. If corresponding node 18 wishes to send a message to Mobile Nodeâ€"whether in reply to a message from the Mobile Node or for any other reasonâ€"it addresses that message to the IP address of Mobile Node 6 on sub-network From its mobility binding table, Home Agent 8 recognizes that Mobile Node 6 is no longer attached to network segment The care-of address may be, for example, the IP address of Foreign Agent Foreign Agent 10 then strips the encapsulation and forwards the message to Mobile Node 6 on sub-network In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network PSTN. A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service. An important component of an H. A gatekeeper acts as the central point for all calls within its zone and provides call control services to registered endpoints. Gatekeepers perform two important call control functions. The second function is bandwidth function. A gatekeeper is not required in an H. However, if a gatekeeper is present, terminals must make use of the services offered by gatekeepers. Gatekeeper functionality may be incorporated into the physical implementation of gateways. An optional, but valuable feature of a gatekeeper is its ability to route H. By routing a call through a gatekeeper, it can be controlled more effectively. Service providers need this ability in order to bill for calls placed through their network. This service can also be used to re-route a call to another endpoint if a called endpoint is unavailable. In addition, a gatekeeper capable of routing H. For instance, if a call to a particular destination phone number is routed through a gatekeeper, that gatekeeper can then re-route the call to one of many gateways based upon some routing logic. In addition, the gatekeeper typically provides an IP address associated with the appropriate H. This gatekeeper will then select an H. However, it is important to note that the distance between the Home Agent and the Foreign Agent may be substantial. Moreover, voice is particularly sensitive to latency. In view of the above, it would be desirable to improve the routing path in order to optimize voice over IP in a Mobile IP environment. This is accomplished, in part, by using a local H. In this manner, the routing path is minimized thereby reducing latency in the voice traffic. In accordance with one aspect of the invention, a Foreign Agent that supports Mobile IP is located on a foreign network and configured to enable a node visiting the Foreign Agent to send IP packets including voice information via an IP address obtained from an H. In accordance with another aspect of the invention, a node visiting a Foreign Agent on a foreign network is capable of sending IP packets including voice information via an IP address obtained from an H. The node receives an agent advertisement identifying an H. The node may then send an IP packet addressed to the IP address and including voice information. It will be obvious, however, to one skilled in the art, that the present invention may be practiced without some or all of these specific details. In other instances, well known process steps have not been described in detail in order not to unnecessarily obscure the present invention. Through connecting to this router , a person may send and receive voice in IP packets through a phone , a personal computer PC or a node such as a mobile node The destination may be a PSTN gateway or another device. In this instance, since the call is being made to the phone , the destination of the IP connection is the PSTN gateway Typically, a node such as a mobile node obtains an IP address associated with a destination from an H. For instance, the H. IP data packets including voice information may then be routed to and from the Foreign Agent via Home Agent In accordance with one embodiment, the Foreign Agent sends an agent advertisement identifying the local H. From this agent advertisement, the mobile node may obtain and save H. The mobile node may then send a packet addressed to the local H. For instance, the destination may be a device or a PSTN gateway. The Foreign Agent then forwards a packet including the requested IP address to the mobile node The IP address is preferably obtained from the local H. However, when the IP address cannot be obtained from the local H. Once the mobile node obtains the IP address, the mobile node may address an IP packet including voice information to the obtained IP address and send the IP packet via the obtained destination address. As described above, the data flow from a mobile node may be optimized through a local H. In addition, data flow may also be optimized on the return path to the mobile node e. The Foreign Agent then forwards these IP data packets to the mobile node Rather than requiring all data packets to be sent via the Home Agent , the corresponding node e. For instance, a packet including the care-of address associated with the Foreign Agent may be sent to the corresponding node e. The corresponding node may then tunnel packets directly to the care-of address. The Foreign Agent may then forward the IP data packets to the mobile node. In this manner, the transmission of IP data packets via the return data path may similarly be optimized. The Foreign Agent advertises a local H. The mobile node receives the agent advertisement at block The mobile node may then save H. The mobile node then sends a packet to the learned H. Thus, it is determined at block whether the learned H. When the local H. Alternatively, when the local H. The mobile node then initiates a call with the obtained IP address at block Once a call is initiated, IP packets including voice information may be sent by the node to its specified destination. Thus, the mobile node sends an IP packet including voice information to the Foreign Agent at block Once a call is initiated, IP data packets may be sent to the mobile node. As shown, at block a packet is sent from the destination to the Home Agent. The Home Agent then tunnels the packet to the Foreign Agent at block The Foreign Agent then forwards the packet to the mobile node at block

Chapter 5 : Voice/Data Integration Technologies - DocWiki

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Chapter 6 : atm and internet protocol | Download eBook pdf, epub, tuebl, mobi

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Chapter 7 : Optimizing Voice In Atm/ip Mobile Networks By Juliet Bates at calendrierdelascience.com

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Chapter 9 : 8 Steps to Optimize Your Network for VoIP | calendrierdelascience.com

networks with a performance management system. The thesis discusses at the general level the most important radio technologies, their requirements and different services they provide.