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ITJ

ISSN 1812-5638

# INFORMATION TECHNOLOGY JOURNAL

ANSInet
Asian Network for Scientific Information
308 Lasani Town, Sargodha Road, Faisalabad - Pakistan

Information Technology Journal 4 (2): 114-117, 2005
ISSN 1812-5638


# Optimal Cost Solutions for Voice Communication on Shared Media

Muhammad Asif Arshad, Muhammad Aslam and Faisal Iqbal
Al Khawarizmi Institute of Computer Science, University of Engineering and Technology, Lahore, Pakistan

**Abstract:** The increase in information services on the shared media such as voice over IP is creating an unprecedented demand for bandwidth. As shared media service providers and backbone service providers continue to upgrade their infrastructure to accommodate this demand, the flat rate pricing schemes and their inherent cross-subsidies will continue to create economic inefficiency and wasteful resource allocation. To pay for the bandwidth that is consumed, IT departments charge costs back to users. This creates some accountability, but the systems that are employed, which are based on flat-rate connection and port charges, are usually too imprecise to truly motivate users to employ shared media more thoughtfully. One solution being implemented is a change in the economic structure of the IP business through usage-based billing. Usage-based billing refers to any billing scheme where a customer's bill is scaled in some way to reflect actual use of network services. A discussion presented in this study based upon some well defined and widely accepted Voice over IP (VoIP) traffic models. Based on arguments for and against usage-based billing, conclusions are drawn regarding the extent to which usage-based billing can be expected to permeate the market.

**Key words:** Voice over IP, cost effective solutions, shared media, usage based billing

## INTRODUCTION

In this study, an optimal cost solution for voice communication on data networks is presented. This concept is based on the assumption that bandwidth utilization is never 100% and if anyone is not utilizing 100% bandwidth, proper rebate should be given for the unutilized bandwidth.

There is exploding growth in data network investment, both by private organizations and by new and traditional carriers. The technology to carry voice over data networks has evolved in the last few years[1]. There are economic advantages to end users in utilizing an integrated network, not only in terms of direct transmission costs, but also in reducing the network management costs of running separate and technologically different networks. That is the ultimate goal. In the meantime, many companies are and will be for some time in the future, supporting the infrastructure and cost of multiple networks including Public Switched Telephone Network (PSTN). Hence the need to optimize the usage of all media components on all networks simultaneously and to take advantage of pricing alternatives between networks will become even more important in the corporate environment and as the service providers offer increasingly competitive prices.

In the traditional utility industry, where individual consumption varies from one customer to another, it is commonly accepted that prices should reflect usage. Yet, in the broadband Internet industry, where individual consumption varies even more, something by several orders of magnitude between customers, the dominating business model is still flat rate pricing. This has to be changed to a model that is based on usage.

## VOICE OVER IP-TRAFFIC MODELS

Packet switched IP networks have established themselves as an attractive option in a wide variety of data communication environment. To the present, circuit switched networks have been the principal mechanism for transmitting human speech on a real time conversational basis. However because of perceived economic and technical benefits, digital voice techniques and corresponding network architectures have received considerable attention in the recent past. These benefits range from noise and cross talk immunity to data and voice compatibility, security, bandwidth conservation, integrated networking, new system synergies and network management cost reductions, among others[2].

Research is underway to develop such design techniques so that with the cost performance trade offs as

**Corresponding Author:** Muhammad Asif Arshad, Research Associate, Al Khawarizmi Institute of Computer Science, University of Engineering and Technology, Lahore, Pakistan
Tel: 042 6827618/26 E-mail: asif@kics.edu.pk

a tool, system optimizations can be performed with the characteristics of the terminals, the network topology and the protocols, all considered jointly. Furthermore assessments can be made of the relative merits of packet switching, circuit switching and hybrid alternatives for handling speed traffic and ultimately for mixed data and speech traffic. Considerations here are restricted to the modeling, analysis and design of packet switching networks carrying only real time speech packets; thus the presence of network control traffic is ignored. Any switch or router design considerations are also ignored and ideal switch or router behavior is assumed.

## TRAFFIC MODEL

The statistical analysis of speech patterns has attracted attention for the past half century. As a product of such study, several models for telephonic speech patterns have emerged. In general, to obtain a better fit to empirically measure data, the models must grow correspondingly in sophistication and complexity. These traffic models are significant because these are environmental conditions that provide reasons why packetized speech network design differs from the data case. In general, speech traffic has a more regular and predictable arrival pattern than data; so one would expect the network design to be able to capitalize on this by achieving higher facility utilization with speech than is possible with data[3].

**Speaker models:** A time slotted environment in which speech takes place is assumed. That is clock divides the time axis into segments, during each of which a speaker generates an empty or nonempty packet, depending on whether the threshold energy was exceeded during that time period. These time segments will be called frames. In the transmitted speech signal, it is assumed that empty packets are non-empty packets. The usual simplifying assumptions, that only one member of the packet listener pair changes speech activity states during a frame is made.

## USAGE BASED BILLING MODEL-SOLUTIONS TAXONOMY

It is generally accepted that WAN bandwidth costs represent both the largest and most rapidly growing cost area with users' network budgets. The taxonomy of solutions for creating accountability with respect to who is spending the WAN budget is as follows.

**Flat rate connection or port charges:** Flat-rate accounting typically involves dividing the cost of network expenditures by the number of employees or departments. Per person accounting is commonplace in the LAN while in the WAN it often happens at the departmental and business unit level. Expenditures may be limited to monthly bandwidth costs or include equipment and services costs[4].

**Home-grown systems:** Network monitoring and performance management software has become fairly standard in corporate WANs. The software compiles data from instrumentation placed throughout the network. Instrumentation may consist of stand-alone probes that monitor WAN and LAN links) and software agents embedded in WAN and LAN equipment (SNMP agents are most common).

The usage reports that monitoring systems produce allow network managers to identify individual users and departments that consume excessive bandwidth and based on this information, modify a flat-rate port or connection system so that these users get charged more than their more responsible brethren[4].

**Usage-based billing systems:** True usage-based billing is a combination of instrumentation and software. Generally speaking there are three elements to these systems: data gathering, data processing and the reporting tool or Graphical User Interface (GUI).

One of the main things that differentiate usage-based billing tools from network monitoring and performance management products is the ability to customize the policy of billing. The policy side represents that ability to charge different prices for time of day use, different applications and even different protocols. Administrators can also choose from a variety of off-the-shelf report formats. One type of report may be distributed to users or departments while a more detailed report may be employed for network planning purposes. The user interface through which all this functionality is accessed is typically a Java application running on a browser[5].

**Policy-based networking:** Policy-based networking has the burden of being the next "big thing" in the networking industry. Policy-based networking provides various levels of service to different users and applications based on various policies that have been assigned. These policies can be either static or dynamic. Lowering the amount of bandwidth allocated to certain users during specific times of the day would be an example of a static policy.

Dynamic policies would allow for streaming video when bandwidth utilization had fallen below a specific percent of capacity. Policy-Based networking will be used initially for two functions: Quality of Service (QoS) and security.

## ANALYSIS OF CALL COSTS USING PTCL AND VOIP

With Old World voice networks, all voice calls use 64 Kbps fixed-bandwidth links regardless of how much of the conversation is speech and how much is silence. With VoIP networks, all conversation and silence is packetized. Using Voice Activity Detection (VAD), packets of silence can be suppressed. VAD is enabled by default for all VoIP calls. VAD reduces the silence in VoIP conversations, but it also provides Comfort-Noise-Generation (CNG). Because one can mistake silence for a disconnected call, CNG provides locally generated white noise so the call appears normally connected to both parties. Instead of sending VoIP packets of silence, VoIP gateways can interleave data traffic with VoIP conversations to more effectively use network bandwidth.

Over time and as an average on a volume of more than 24 calls, VAD may provide up to a 35% bandwidth savings. The savings may not be realized on every individual voice call, or on any specific point measurement. But savings significance can be judged by taking into account PTCL tariffs and comparing these tariffs with VoIP call charges.

Table 1 shows some combinations of codec, payload size, Real-Time Transport Protocol (RTP) header compression and Voice Activity Detection (VAD). RTP header compression is also referred to as compressed Real-Time Transport Protocol (cRTP)

The following assumptions are made:

- The IP/User Datagram Protocol(UDP)/RTP headers are 40 bytes.
- RTP header compression reduces the IP/UDP/RTP headers to 2 or 4 bytes.
- Multilink Point-to-Point Protocol (MLPPP) or Frame Relay Forum (FRF).12 adds 6 bytes of layer 2 header.

The following calculations were used from Table 1:

- Voice packet size = (layer 2 MLPPP or FRF.12 header)+(IP/UDP/RTP Header)+(voice payload)
- Voice packets per second (PPS) = codec bit rate/voice payload size
- Bandwidth = voice packet size * PPS

Table 1: Basic information of VOIP calls

| | |
|---|---|
| Compression technique codec bit rate | G.729 (8) |
| Payload size | 20.0 |
| Bandwidth MLPPP or FRF.12 | 26.4 |
| Bandwidth w/cRTP MLPPP or FRF.12 | 11.2 |
| Bandwidth w/VAD MLPPP or FRF.12 | 17.2 |
| Bandwidth w/cRTP and VAD MLPPP or FRF.12 | 7.3 |

Table 2: Set 1 of VoIP calls

| Call No. | Call duration (Sec) | VoIP packets | Actual duration (Sec) |
|---|---|---|---|
| 1 | 30 | 1125 | 22.5 |
| 2 | 60 | 2688 | 54.0 |
| 3 | 180 | 6923 | 139.0 |
| 4 | 300 | 14250 | 285.0 |
| 5 | 600 | 27521 | 551.0 |

Table 3: Set 2 of VoIP calls

| Call No. | Call duration (Sec) | VoIP packets | Actual duration (Sec) |
|---|---|---|---|
| 1 | 30 | 1427 | 29 |
| 2 | 60 | 2949 | 59 |
| 3 | 180 | 7567 | 152 |
| 4 | 300 | 14579 | 292 |
| 5 | 600 | 25739 | 514 |

For example, the required bandwidth for a G.729 call (8 Kbps codec bit rate) with cRTP, MLPPP and the default 20 bytes of voice payload is:

- Voice packet size (bytes) = (MLPPP header of 6 bytes)+( compressed IP/UDP/RTP header of 2 bytes)+(voice payload of 20 bytes) = 28 bytes
- Voice packet size (bits) = (28 bytes) * 8 bits per byte = 224 bits
- Voice PPS = (8Kbps codec Bit Rate )/(160 bits) = 50 PPS Note: 160 bits = 20 bytes (default voice payload) * 8 (bits per byte)
- Bandwidth per call = voice packet size (224 bits) * 50 PPS = 11.2 Kbps

To give the practical significance of the usage based billing technique developed, we take example of two sets of three calls (Table 2 and 3). The VoIP calls were made possible with the co-operation of Corvit Systems, Lahore.

The time required to send the packets on the shared media will be less than the total time of the call. Finding the total number of bits to be transferred and dividing this number by the rate at which transfer takes place can be used to calculate this time.

The gap between the cost of PTCL and VoIP call increase manifold as the duration of the call increases. VoIP tends to get cheaper as the call duration increases. This cost of the VoIP call also includes fixed charges for occupying the communication channel of the telecommunication company. This fixed rate has been taken as additional 50% of the bandwidth charges.

## CONCLUSIONS

As the telecommunication market opens up in Pakistan and more companies come into business, the competition will grow. So a company providing its users a more accurate billing methodology will certainly get an edge over others. No two persons have same bandwidth utilization in equal amount of time. So the rule should be "More You Use the Bandwidth, More You Pay". Proper rebate should be given to the user who is not using that much bandwidth.

Just like telephone, gas and electricity, bandwidth is a measurable commodity. The more customers a provider has, the lower the per user costs for distribution infrastructure, therefore the cost of providing services for the broadband service provider.

Cost-justifying a usage-based billing system involves determining the initial rupee outlay required for the system and the payback period given projected bandwidth savings. Bandwidth costs in any given corporation's network may be growing faster or slower than the assumptions used. The trend is that many corporations are experiencing much higher growth rates. As for the assumptions regarding bandwidth reduction, it is hard to generalize about their applicability to the industry as a whole. But usage-based billing has not been around long enough nor has it been implemented by enough users to draw firm conclusions as to whether the assumption is conservative or optimistic. Other variables that affect cost include the extent to which usage-based billing is to be employed. Many corporations will doubtlessly pick and choose what to charge for to a greater degree.

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