Voice traffic over Internet versus ordinary trunks

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To My Parent and My Wife

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ABSTRACT

This dissertation is aimed to study the introduction of VoIP trunks instead of the ordinary leased trunks. The dissertation considers, for the case study, two of the Sudan Telecommunication Company pilot projects executed mainly to study the impact of introducing voice over the Internet in the international call trunks. The dissertation addresses the general technical issues in applying VoIP and brief analysis on the performance parameters. The devices implemented, configurations deployed and the connectivity are demonstrated. The designs of the cases with the capacity of resources used are verified. The transported packets are captured and decoded. The carried traffics are analyzed and compared with ordinary trunk traffic.

The results of this study show that the implemented VoIP trunks has carried a total volume of incoming-traffic of 130% of that carried by the ordinary trunks. The incoming volume of traffic via the ordinary trunks is dropped to 70% of traffic carried before introducing VoIP trunks. The implemented solution demonstrates that the deployment of VoIP trunk does not need establishment or reconfiguration of the international link, not a big platform change, and not very complicated devices to be used.

Delay measurements of VoIP trunk show that the delay is within the recommended range.

Statistical analysis of the traffic shows that VoIP has similar daily profile as that of the ordinary voice trunks, which will not affect the current call pricing policy. The traffic load pattern of VoIP has shown that there is always a room for summing up trunks to maximize the circuit utilization.

Calculation shows that VoIP trunks obtain better answer-to-seizure ratio and higher average holding-time than that of the ordinary one, thus more paid time than that of the ordinary one is expected.

Measurements of IP band width and calculation of the carried traffic volume estimated that if one fourth of the capacity of the current ordinary voice trunks is changed into VoIP trunks, then the traffic volume is expected to be doubled.

بسم ا   الر ن الرحيم

مستخلص

الهدف من هذا البحث إجراء دراسة عن تطبيق تقنيات الاتصالات الحديثة لنقل الإشارات الصوتية داخل حزم البيانات. ولقد ركزت الدراسة على استخدام تقنية حمل الصوت على البروتوكول(IP) عبر شبكة الإنترنت لنقل المحادثات الهاتفية العالمية بديلاً عن نقلها بواسطة دوائر الهاتف المخصصة والعاملة حالياًعلى تقنية الصوت التقليدية.

أُجريت الدراسة على نموذجين يجرى تقييمهما حالياً بواسطة الشركة السودانية للإتصالات كمشروعين للإستدلال بنتائجهما في إعتماد هذه التقنية. غطت الدراسة النواحي العلمية والفنية حول تطبيقات هذه التقنية. وأُجريت خلالها قياسات عملية لعوامل الأداء. كما شملت تقييم حسابي للتصميم المقدم وتحليل إحصائي للحركة المحمولة.

تناول الجانب العملي فيها فحص لطبوغرافية التوصيل وإعداد الأجهزة. وكذلك مراقبة البيانات المرسلة وتفسيرها بواسطة أجهزة تحليل للبيانات. وأيضاً تجميع بيانات الحركة من داخل الأجهزة ومقسمات الهاتف ثم قياس عوامل الأداء من بطء واستغلالية لسعة الربط.

إتضح من نتائج الدراسة أن حجم الحركة الواردة بدخول هذه الدوائر قد بلغ 130% من حجم الحركة عبر الدوائر التقليدية. كما نقص حجم الحركة المنقولة عبر الدوائر التقليدية الحالية إلى 30% من حجم حركتها سابقاً. إتضح من فحص الأجهزة أن إدخال هذه التقنية في أنظمة الدوائر العالمية لا يحتاج إلى تغيير جذري في البنية التحتية للشبكة المستخدمة حالياً أو إلى إنشاء دوائر أتصال جديدة.

قياس زمن تأخير التراسل عبرهذه الدوائر يوضح أن زمن التأخير يقع تماماً في المدى الموصى به في المعايير القياسية.

التحليل الإحصائي للحركة المنقولة أوضح أن نمط الحركة لمثل هذه الدوائر وتغييره اليومي يشابه إلى حد كبير نمط الدوائر التقليدية مما يجعل دخول هذا النوع من الدوائر غير مؤثر على سياسة التسعيرة المتبعة حالياً في الدوائر التقليدية. كذلك تلاحظ من تباين نمط الحركة الاسبوعي

والشهري أن هناك إمكانية لدمج مجموعة من الدوائر في سعة ربط واحدة مشتركة بينهمـا ممـا يحسن مستوى استغلالية الدائرة.

حساب معدل إشغال الدوائر ومتوسط زمن المحادثات أظهر أن القيم في هذه الدوائر عالية مقارنةً بالدوائر التقليدية، مما ينبئ بزيادة الزمن المدفوع في المحادثات لهذه الدوائر عن نظيراتهـا مـن الدوائر التقليدية.

حسب نتائج القياسات التي أُجريت على استقلالية سعة دوائر الإنترنت، وحساب حجـم الحركـة الهاتفية المحمولة بواسطة هذا النوع من الدوائر فيمكن التوقع بمضاعفة حجم الحركة إلى الضعف متى ماتم تغيير ربع السعة المستخدمة حالياً في الدوائر العالمية لتعمل على هذا النوع من التقنية.

1 INTRODUCTION

People communicate between each other using speech, which in nature is a form of voice. Voice is defined as a mechanical vibration in a wave pattern of 4kHz bandwidth. These waves when communicated to the ear it is then transmitted by the nervous system to the brain where the sensation that understood as voice is experienced.

Mankind since the early life is looking forward to transport this speech beyond the audibility limit to a far distance listener. Many means were used to communicate between far a part people in a form of signs like using fire, smoke or beats of a big drum. Written language used to be transported physically in a paper-like form. But all these means were not able to transport the spoken speech in a real time to a remote listener.

After the invention of electricity, the voice wave was changed into an electrical signal. Microphones translate the vibration waves into electrical signals, and loudspeakers retranslate it back into sound. The continuous electrical signal, analogue signal, could be transmitted over wires only for hundreds of meters. Thus the speech could be transported in a real time but for a very limited distance.

When telegraph was invented, the text rather than the speech could be transported to more far distance. Alphanumeric letters were encoded in bits and thus transported in a form of electrical pulses over the wire for several kilometers.

Later on, Radio waves were used to carry the voice signal. Radio wave is capable to propagate for very long distances. Many researches done on modulating Radio carriers with the voice signal to achieve transmission of speech in a real time for a very long distance. Analogue signal that carried in Radio has a high noise level, high power for signal generation, and has a limited range of frequency. Radio was able to transport the speech in a real time, but it is costly or limited in frequency and distance ranges.

Nyquist, in his theorem states that samples of a continuous signal can reconstruct its original signal if the sampling rate is greater than twice the signal frequency. This theorem opens the door to digitize the voice. Digitization encode the level of signal at each sampling event by a number of bits, the analogue voice signal is thus changed into a stream of pulses representing the original signal. Digital signals can be regenerated many times, enabling it to be transmitted for long distances using digital transmission.

Long distance channels are very expensive, thus many techniques were developed to fully utilize the link by multiplexing many conversations of speaking parties into one channel. Researches focus on mathematical modeling of voice signals/ vocal tract and on studying the nature of human sensation organism. The mathematical properties of the modeled signal help in coding samples of the signal, the difference in change, or prediction of the original signal. Thus the amount of bits required to code the signal is minimized. Coding algorithms of less number of bits are the much complex, and take a longtime in digital processing for encoding and decoding the voice signal, thus causing a recognized delay in speech.

Enhancements in digital processors and distributed digital processing techniques enabled industry to develop digital signal processors (DSP), which are widely used for real-time processing of voice signals. The digital signal processors minimize the delay in processing and provide a possibility for more digital processing like packetization.

Packetization of voice bit stream enables it to be transported over packet switching networks. Packet switching networks will become the main networks in telecommunications. The encoding and multiplexing it uses are highly efficient in circuit utilization, but it suffers some processing delay and packets loss. The real-time speech is very sensitive to these effects. Many mechanisms and algorithm are developed to minimize these effects.

More sophisticated predictive algorithms are used to code the voice signal in a very few number of bits. Techniques to suppress the silence in speech are also used. Reliable protocols to guarantee the packets transportation are used and a dynamic buffer that controls any variation in delay is used as well. Thus speech can be transmitted real time for long distances over packets based networks.

The rapid increase in the deployment of IP-based networks to face the growth in data transmission applications and needs of Internet users, drives researches to introduce techniques for transporting more telecommunication services over IP-based networks. Many techniques succeeded in transporting the telephone calls over IP-based networks with acceptable level of speech quality and solution of security issues.

The transmission of voice over IP (VoIP) leads to convergence of the data networks and the telephone networks in one network and platform. The telecom operators will then need to migrate from the legacy networks architecture to a next generation of networks (NGN). The International Telecommunication Union (ITU) has recommended a general strategy to migrate to NGN as a safe approach, technically and economically.

One of the first steps for migrating to NGN is the voice over packet trunks. The telecom operators' start the migration by replacing the ordinary leased voice trunks for the long distance calls by voice over IP trunks. VoIP trunks highly reduce the cost per call in capital and operation expenses.

The dissertation addresses the general technical issues in applying VoIP and brief analysis on the performance parameters of VoIP trunks versus that of the ordinary trunks. In the next part, Literature Review, the main technologies that lead to the introduction of VoIP are briefly described. The deployment scenarios as well as processes and techniques applied are explained. The performance parameters with their standards and recommended ranges are highlighted too.

The third part of this dissertation, Experimental Measurements, describes the experiments and test benches conducted for capturing and collecting the performance parameters. Equipment and analysis tools used in the experiment with their connection topology are illustrated. The measurement methodology is explained as well.

The fourth part, Results and Calculations, shows the calculation and statistical analysis done on the collected data. Results of performance parameters are identified, and characteristics of traffic patterns are classified.

The last part, Conclusions, discusses findings came out of the analysis, and points out suggested solutions and expected benefits.

The collected data of measurements done in the dissertation are tabulated and contained in the enclosed compact disk (CD). Samples of these tables are shown on the thesis. Complete tables of results and sub-tables of intermediate calculations that carried out during the analysis are contained in the CD also.

2 LITERATURE REVIEW

Telecommunication networks are established to provide connectivity between any two points that are a distance apart in order to communicate between each other. They are principally constructed by using a set of infrastructures that are shared by subscribers in order to reduce the cost. There are two types of telecommunication networks. One is used for voice communication, which is the Public Switching Telephone Network (PSTN), and the other one is the Public Data Network, which is established and optimized mainly to transport data.

2.1 Legacy telephony network architectures

Telephone networks were established basically to provide a universal voice communication service with a good quality. The technology used in telephony networks is known as circuit switching. It is based on the principle that a circuit must be reserved for a call from the time that it is set up to the time that it is ended.

In circuit switching, a dedicated path is established between the two stations for communication. The switching and transmission resources within the entire network are reserved for the use of this call. All of the channel capacity is dedicated for this connection, while the actual utilization of the voice connection itself doesn't reach 100%; the circuit switching is thus rather inefficient in using these resources [1].

Voice communication between any two parties is achieved by transmitting control signals from the source to the destination through the entire network. Calls are established, maintained and terminated by using these signals. The most widely used signaling protocol in such network is Signaling System Number 7 (SS7).

SS7 is an open-ended signaling standard covers all aspects of control signaling for complex digital switching networks. The control messages are routed through the network to perform call management (setup, maintenance and termination), and to perform network management function as well [2].

The switching nodes, in such networks, are connected to one another by some kind of transmission channels (trunks) that carry multiple voice circuits. Transmission sources share a large transmission capacity by some form of multiplexing. The common application of multiplexing is in long communication links, like fiber, coaxial or microwave links, which can carry large numbers of voice and data messages simultaneously using multiplexing. The type of multiplexing technology used in circuit switching is the time-division multiplexing (TDM).

TDM multiplexing carries data from various sources in repetitive frames, each frame consists of a set of time slots and each source is assigned one or more time slots. The time slot is dedicated for each source even if there is no data transmitted; thus the TDM multiplexing is an inefficient multiplexing technique.

2.2 Data network architectures

In a communication between any two computers, the line is idle for most of the time; thus circuit switching is not efficient for such application. The data networks are then established to interconnect computer networks in a more efficient way. Data networks support applications that transfer data in packets, from sources to their destinations with an agreed quality of service. The current technology used for this purpose is known as packet switching.

Packet switching transmits the data in short packets, each packet contains the user's data plus some control information required by switches in the network to route and to deliver the packet to the intended destination. At each node in the route, the packet is received, stored briefly for switching process, and then forwarded to the next node according to the destination address. Therefore the packet, in this technology and relative to circuit switching, experiences some delay in each node.

Instead of the circuit switching signaling protocols, the nodes in these networks use some routing or switching protocols between each other to discover the entire network or the neighbor node to where the packet should be forwarded.

The multiplexing technology used in these networks is known as statistical time-division multiplexing, where the time slots are not pre-assigned to particular data sources, but the data is buffered and transmitted as soon there is an available time slot. Time slots are dynamically shared according to the incoming bit rate statistics. Statistical time-division is a more efficient form of multiplexing than synchronous TDM, specifically in transporting data that is not sensitive to delay.

2.3 Internet and protocols stack

The Internet is the largest example of data networks now deployed on Earth. It transfers the data on a "best effort" basis, and if data is corrupted it is resent to its recipient without any other constraints. The Internet is based on a packet switching protocol known as the Internet Protocol (IP).

IP protocol is widely used in data networks. It routes information across interconnected networks by transferring data packets from one IP device to another. The data packets are transferred from the source to the destination according to an interpretation of an address. The IP protocol forms part of layer 3 of the TCP/IP model, it is a simple protocol without any error control.

Thus the IP protocol is not reliable; the packet transmission is controlled by the higher-layer transport control protocol (TCP). It is a reliable protocol that corrects the errors of the underlying IP protocol. Also, the TCP header of each packet contains its sequence number that helps to put the data streams back in order at the receiving terminal. When a packet is repeatedly lost, its recovery will lead to a significant time lag. Since audio and video applications involve constant throughputs that can be interrupted by such time lag and delay variations, then the TCP protocol is unsuitable for this type of application beyond a 4 or 5% packet loss rate [3].

In these types of applications, continuity is more important than reliability. It is thus the User Datagram Protocol (UDP) that is used rather than the TCP protocol. The UDP protocol is a connectionless oriented protocol. It is an unreliable protocol without any error correction. Therefore a more reliable protocol, Real-time Transport Protocol (RTP), is employed above the UDP datagram.

The RTP is a transport and control protocol meant for the transport of real-time data. It provides sequential reconstruction, packet loss detection and data content identification. The RTP header contains information for synchronization and recovery of the signal, a time stamp, and stream and sequencing indices. The RTP has no congestion or transmission delay control mechanisms. RTP is generally used with the Real-time Transport Control Protocol (RTCP) for transmission control.

The RTCP protocol transmits periodic control packets to all participants in a session, inform the sender of the transmission quality, packet losses and delays, so as to modify the transmission accordingly.

When a certain service or type of data is needed to be transferred through the Internet with special considerations, then a field "Type of Service" and a priority bit in the IP datagram are used to mark and classify each packet for the desired services.

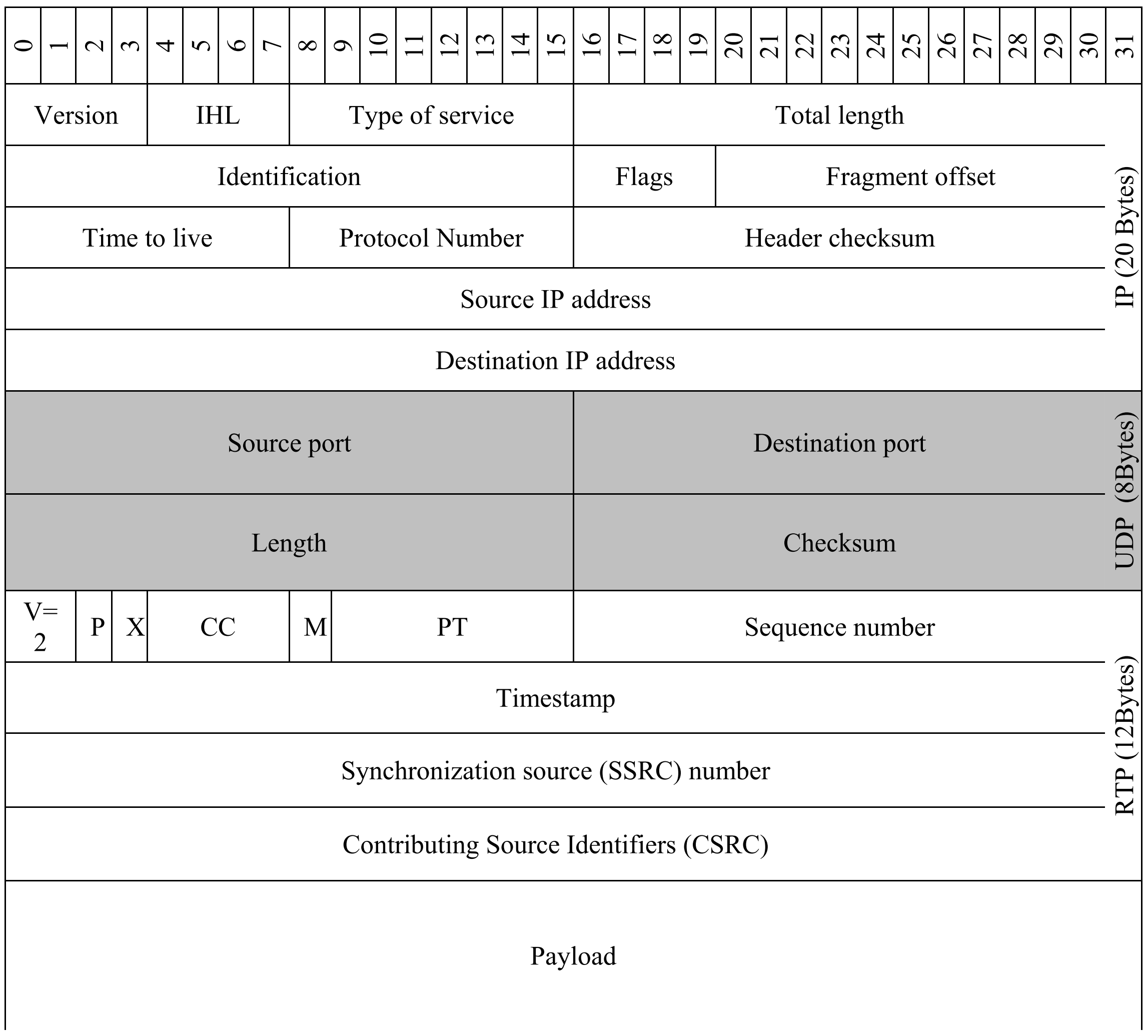


Figure (2.3): Protocol Stack of VoIP Packet

2.4 Coding technologies in VoIP

The IP transmission channel has a considerable variation in delay that degrades the quality of voice in VoIP applications. To recover from such undesirable effects, some efficient coding techniques have been developed and are being used in transmitting audio-video over IP networks. The speech encoders used in VoIP applications employ Code Excited Linear Predictive (CELP) coding, which is defined by the ITU Recommendation G.729 [4]. It is based on analysis-synthesis techniques.

Synthesis coding processes blocks of samples of the voice signal and constructs a model that generates samples, which are statistically identical to the original signal samples. The excitation function, which is a linear combination of random and periodic functions, obtains good quality of the voice at bit rates between 5 and 16 Kbps. Each G.729 call uses a bandwidth of 26.4 Kbps [3].

2.5 Voice activity detection (VAD)

Measurements of voice signal activity show that 40 -50% of the time there is no speech content in one direction of the channel. Voice activity detection (VAD) technique is used to ensure that the channel is used by a signal only when there is actual voice to be transmitted.

In VoIP, when the G.729 coding technique, compressed RTP header (cRTP) and voice detection activity (VAD) are used then the IP bandwidth consumed by each call is only 9.6 Kbps.

2.6 VoIP Scenario

There are three VoIP usage scenarios according to the network connections and the type of the terminal device used.

i) PC-to-PC Scenario: In this scenario the call is established between two computers that are connected to the Internet via an Internet service provider (ISP). The two parties have to be connected to an online directory server where they register prior to each communication. They establish the voice communication by using VoIP-compatible software. The protocols used between the two communicating parties are the ITU-T H.323 protocols.

ii) Phone-to-Phone Scenario: In this scenario the calling and the called parties are the ordinary telephones, which are connected to the public telephony network. They establish a voice communication by either a use of gateways, or by use of adapter boxes. VoIP gateways are used by the telephony providers to transmit the voice calls over a managed IP network with acceptable quality of service. This is used internally between switches instead of the ordinary trunks. This implementation is transparent to telephone users.

Some special modem boxes are installed between the user's telephone set and his connection to the PSTN. Each user needs to have a subscription with an ISP whose access parameters are preprogrammed in the box. The calling party initiates a call in the same way as in PSTN. In the first phase the modem dials to the ISP and then it establishes an IP connection between each of the two correspondents [3].

iii) PC-to-Phone or Phone-to-PC Scenario: In this scenario, one of the call parties has a computer connected to the Internet via an ISP, while the other is an ordinary telephone subscriber. For the party with a PC to place a call to a telephone set, first a connection with the Internet is established and then the call is established via an Internet telephony service provider (ITSP), which operates a VoIP gateway that accesses the PSTN of the called telephone. In the reverse turn, the ordinary telephone places a call to a PC by dialing an E.164 number that mapped to an IP address allocated by the IP telephony operator [4].

2.7 Voice over Internet processing

The internal structure of a VoIP gateway contains the followings:

- Digital Signal Processor (DSP) which captures the voice packets and carries out the required real time processes like, voice compression, tone detection and generation, echo cancellation, silence suppression, and control signals translation.
- Microprocessor, which performs operations of the telephony protocols, network protocols, routing algorithms, device management and billing records.

1. The VoIP call over Internet under goes the following processes [5]:
2. The received analog voice signal is converted to a Pulse Code Modulation (PCM) digital stream.

3. The PCM stream is then analyzed, the echo is removed, and the voice activity detector (VAD) removes the silence. Tone is detected and routed around the CODEC.
4. Remaining PCM stream are forwarded to the CODEC for compression. The voice frames are then created with equal size and generated at a constant rate while someone is speaking.
5. The packet assembler software within the DSP takes frames from the CODEC and creates the packets. A 12-byte Real Time Protocol (RTP) header is added providing sequence number and time stamp.
6. The packet is forwarded to the gateway's host processor for addressing. The dialed digits identified by the tone detection performed in the DSP are used to determine the destination number and then it is mapped to an IP Address.
7. An 8-byte UDP header containing source and destination port/socket addresses is also added.
8. A 20-byte IP header is added to the packet containing the IP addresses of this gateway and the destination gateway.
9. Through the Internet, routers and switches examine the IP addresses to take packets to their destination. Packets may be delayed, disordered or lost.
10. In the destination an adaptive jitter buffer in the receiving DSP is used to smooth out the delay of packets faced by a jitter.
11. An algorithm in the DSP detects missing packets and replays the last successfully received packet.
12. The DSP detects an out of order condition, the missed packet is then replaced with the predecessor packet, and when the late packet finally arrives it is discarded.
13. The Microprocessor removes the IP and UDP headers from the packet.
14. Packets are then forwarded to the DSP to remove the RTP header.
15. Finally, the packets are disassembled leaving the voice frames.

2.8 Quality of service in transporting voice over Internet

When transporting voice over the Internet, the overall voice quality is a function of many factors including the compression algorithm, frames errors, frame loss, echo cancellation, and delay.

2.8.1 Delay

There are two types of delay, fixed and variable. The fixed delay is due to processing time in each device in the network. The variable delays, which are called jitter, are caused by queuing delays in the egress buffers on each device in the network. The fixed and variable delay sources in the network include:

2.8.1.1 Coder delay (χ):

Which is the summation of:

- The compression delay, which is the time taken by the digital signal processor (DSP) to compress a block of PCM samples. It varies according to the coding and DSP used. G.729 coding technique has a worst compression delay of 10milliseconds [6].
- The decompression delay, which is the time taken to decompress a block of samples, it is roughly 10% of the compression time for each block.
- The compression algorithm delay, which is the time taken to reproduce the sample block. The algorithmic delay for G.729 coder is negligible.

2.8.1.2 Packetization delay (π):

It is the time taken to fill a packet payload with an encoded and compressed speech. It is a function of the sample block size and the number of blocks placed in a single frame. The maximum packetization delay of G.729 coding technique is 30ms. Packetization delay and algorithmic delay are normally overlapped in processing.

2.8.1.3 Serialization delay (σ):

It is the time required by the device for clocking a frame onto the network interface. It is a fixed delay that depends on the clock rate at the interface and on the frame size. The serialization delay required for a 64-bytes frame at a line speed of 256 Kbps is 2 ms.

2.8.1.4 Queuing/Buffering delay (β):

It is the time taken buffers by a frame waiting for routing till its transmission. It is a variable delay that depends on the trunk speed and the state of the queue.

2.8.1.5 Network switching delay (ω):

It is the time taken to transport a frame through the intermediate public data network, which is the source of the largest delay and difficult to be quantified. It is composed of the fixed signal propagation delays. The estimated propagation delay in ITU-T G.114 Recommendation is of 6 ms/km. In general the delay in public data networks for a worst case is 65 ms, 40 ms fixed and 25 ms variable.

2.8.1.6 De-jitter delay (Δ):

It is the time taken by de-jitter buffer to transform the variable delay in the speech into a fixed delay. The initial send-out delay for de-jitter buffer is configured to be equal to the total variable delay along the connection. The maximum depth of the buffer is set to 1.5 or 2.0 times this value.

For the total delay in a good quality of voice connection the accepted delay limit is 250 ms one-way. If delays rise over this limit, the talkers and listeners go out of synchronization. The ITU-T G.114 recommends one-way delay limits for voice applications in three defined bands of delay as shown in the table below.

Range in Milliseconds	**Description**
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrators are aware of the transmission time and it's impact on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes, however, it is recognized that in some exceptional cases this limit will be exceeded.

Table (2.8.1.1): ITU-T Recommended Delay

Therefore the total delay of any two hops transferring voice over a public network is arranged so as not to exceed the limits shown in the following table.

Delay Type	**Fixed (ms)**	**Variable (ms)**
Coder Delay, χ_1	18	
Packetization Delay, π_1	30	
Queuing/Buffering, β_1		8
Serialization Delay (64 kbps), σ_1	05	
Network Delay (Public Frame), ω_1	40	25
De-jitter Buffer Delay, Δ_1	40	
Coder Delay, χ_2	15	
Packetization Delay, π_2	30	
Queuing/Buffering, β_2		0.1
Serialization Delay (2Mbps), σ_2	0.1	
Network Delay (Public Frame), ω_2	40	25
De-jitter Buffer Delay, Δ_2	40	
Totals	258.1	58.1

Table (2.8.1.2): The Total One-way Delay Recommended for Good Quality

2.8.2 Packet loss

Packet loss results in missing part of the speech when the audio signal is received. The voice quality depends on the number of packets lost. There are four main causes of packet loss:

- Exhaustion of the lifetime of the packet (TTL = 0).
- The delay at the receiving end is greater than the jitter buffer.
- Destruction of the packet by any congested devices.
- Packet invalidity caused by faults or errors in the transmission.

The rate of packet loss depends on the quality of the lines used and on the numbers of node in-between. The acceptable rate of the lost packets is less than 20% of the total transmitted packets per call.

2.8.3 Echo

In VoIP the echo, which is the returned part of the processed signal produced by the analogue components, is canceled by using high-performance echo chancellors equipped at the gateway stage of the network.

2.9 Traffic analysis

The traffic analysis is used to engineer and size the network. A well-engineered network has a high circuit utilization and low blocking. Traffic theory formulae are used to measure traffic load. The objective of the teletraffic theory is "*to make the traffic measurable in well-designed units through mathematical models and to derive the relationship between grade-of-service and system capacity in such a way that the theory becomes a tool by which investments can be planned*" [7].

2.9.1 Traffic intensity

The traffic intensity, as in ITU-T E.600 Recommendations, is defined, as *"The instantaneous traffic in a pool of resources, is the number of busy resources at a given instant of time*". The unit used for traffic intensity is Erlang (E), which is the traffic intensity in a pool of resources when just one of the resources is busy. Traffic intensity is equivalent to the product of call arrival rate and the mean holding time.

$$\text{Traffic intensity, during time interval } t = (\text{calls arrival rate})(\text{call mean holding time})$$

$$= \left(\frac{\text{number of calls arrived}}{\text{observation time}}\right)\left(\frac{\text{total seizure time}}{\text{number of calls arrived}}\right)$$

$$= \left(\frac{\text{total seizure time}}{\text{observation time}}\right)$$

Traffic volume in a given time is equivalent to the sum of the holding times in the given time interval. The unit used for traffic volume is Erlang Hour (Eh).

$$\text{Traffic volume} = (\text{traffic intensity})(\text{observation time})$$

2.9.2 Sampling methods

To accurately assess the traffic many samples of the offered load are taken for analysis. The ITU-T recommends measurement or read-out periods of 60 minutes and/or 15-minute intervals.

2.9.3 Traffic modeling

The most adopted traffic models in teletraffic are Erlang B, Extended Erlang B, Erlang C, Engset, Poisson, EART/EARC, and Neal-Wilkerson. The appropriate traffic model is chosen with the consideration of the following criteria:

(i) The call arrival pattern, whether it is a smooth, peak or random pattern.

(ii) The blocked call, whether it is held, cleared, delayed or retried.

(iii) The number of sources, to size the trunks for a given grade of service.

(iv) The holding times, whether it is constant or exponential.

2.9.4 Traffic analysis of VoIP networks

VoIP gateway has an IP leg and a voice leg. Traffic analysis in the IP leg considers the bandwidth to the IP network; while in the voice leg it considers the number of voice trunks.

2.9.4.1 IP leg

VoIP traffic uses Real-time Transport Protocol (RTP) to transport voice traffic. The voice codecs, the samples, the voice activity detection, the RTP header compression and the connection topology affect the bandwidth.

Voice codecs impact bandwidth because they determine the payload size of the transferred packets. The number of samples per packet is another factor in determining the bandwidth of a voice call. Typical voice conversations can contain up to 35 to 50 percent silence, therefore voice activity detection (VAD) is assumed to reduce the bandwidth for each call by 35 percent. The 20 bytes of IP/UDP/RTP headers take up a considerable amount of overhead. By using RTP header compression (cRTP), these headers can be compressed to 2 or 4 bytes offering substantial VoIP bandwidth savings.

2.9.4.2 Voice leg

The traffic model used in VoIP is always Erlang B model [8]. This model is based on the assumptions that the number of sources is infinite, the traffic arrival pattern is a random, the blocked calls are cleared, and the hold times are exponentially distributed.

Erlang B traffic model can be found by the following formula:

$$\mathrm{B}(c,a)=\frac{\dfrac{a^{c}}{c!}}{\sum_{k=0}^{c}\dfrac{a^{k}}{k!}}$$

Where:

B (c, a) is the probability of blocking the call.

c is the number of circuits, ***a*** is the offered traffic.

3 EXPERIMENTAL MEASUREMENTS

The case study for this dissertation was selected to be two of Sudan Telecommunications Company "SUDATEL" pilot projects, which are formed on an objective of introducing the VoIP technology to transport International call instead of using the ordinary leased voice trunks. SUDATEL has made deals with E-Z Connections Inc. (EZC) and ITC Co. to establish voice trunks over the Internet between USA and Sudan for the incoming calls only.

The experimental measurements are to carry out technical evaluation and cost analysis for the solutions implemented in these projects. The results are to be compared with the International standards, as well as with that of an ordinary voice trunk, which is currently connected to USA PSTN via AT&T Co.

The transported packets were captured and then decoded to verify the protocol stack. The IP bandwidth and voice trunks were proved for the optimum design. The current end-to-end delay was measured and compared with the recommended tolerances. Finally, traffic analysis was done to model the traffic pattern and to compare the traffic load carried by the two types of trunks.

3.1 Description of the case study

3.1.1 Logical topology

The project solutions implement a VoIP voice trunk between a USA telephone operator and Sudan PSTN connected through the Internet. In EZC solution the IP bandwidth is 256Kbps and the voice circuits are 30 channels. In ITC solution the IP bandwidth is 512Kbps and the voice circuits are 120 channels. VoIP gateways are installed and connected to the PSTN at each end. The logical connection topology is as shown in Figure (3.1.1).

3.1.2 Devices connections

In the case study the VoIP gateways are connected to Sudan PSTN by direct E1 G.703 interfaces to Khartoum-South local exchange, which is a digital telephone switch, using R2 signaling control protocol. On the IP side the gateways are connected to the Internet by Ethernet (10BaseTX) interfaces using IP routers. The physical connection is as illustrated in Figure (3.1.2).

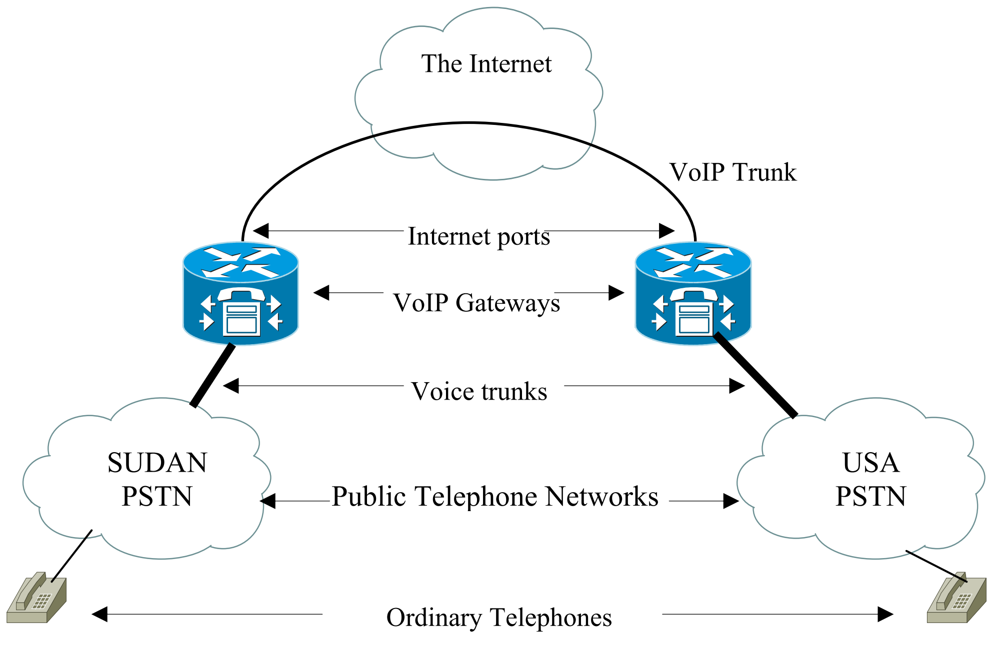


Figure (3.1.1): Logical Connection Topology

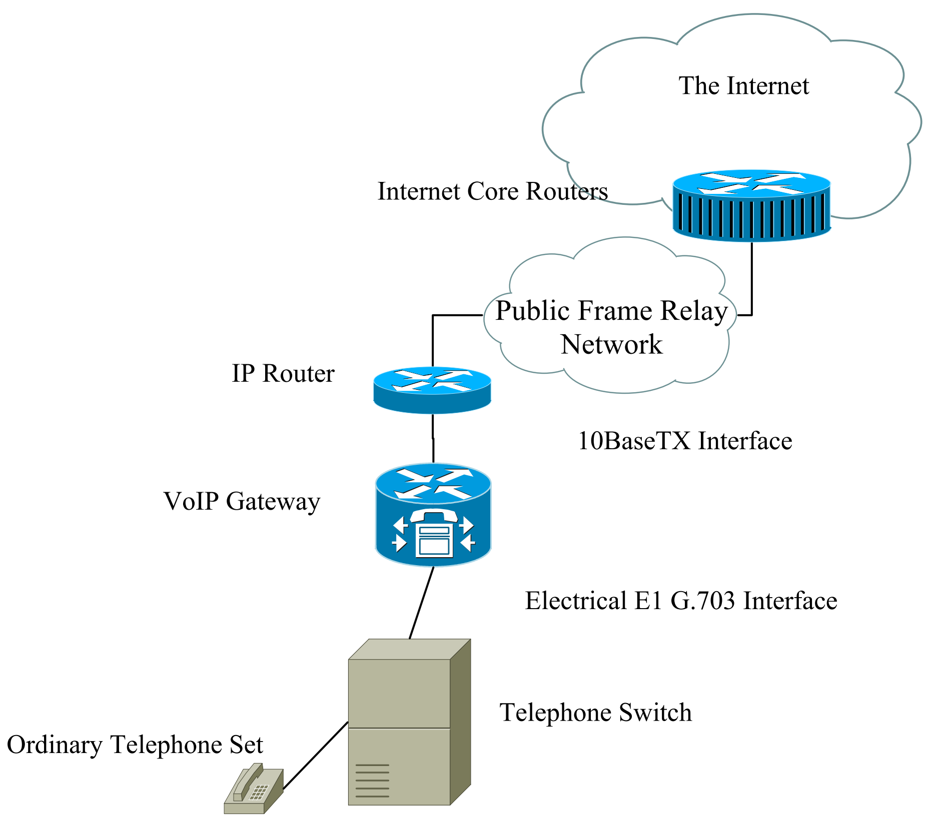


Figure (3.1.2): Physical Connection of the Devices

3.2 Types of the collected data and the collection methods

The operation data such as the traffic statistics, performance parameters and transported packets were captured, collected and analyzed.

3.2.1 Protocols stack

To build the protocols stack of the transported packets, a protocol analyzer was used. Packets were captured using an internetworking analyzer (Wandel&Goltermann DominoWAN DA-310 Internetwork Analyzer). The captured packets were then analyzed using a real time protocols analyzer (ACTERNA DominoNAS). This monitoring bench was set up for EZC solution as shown in figure (3.2.1).

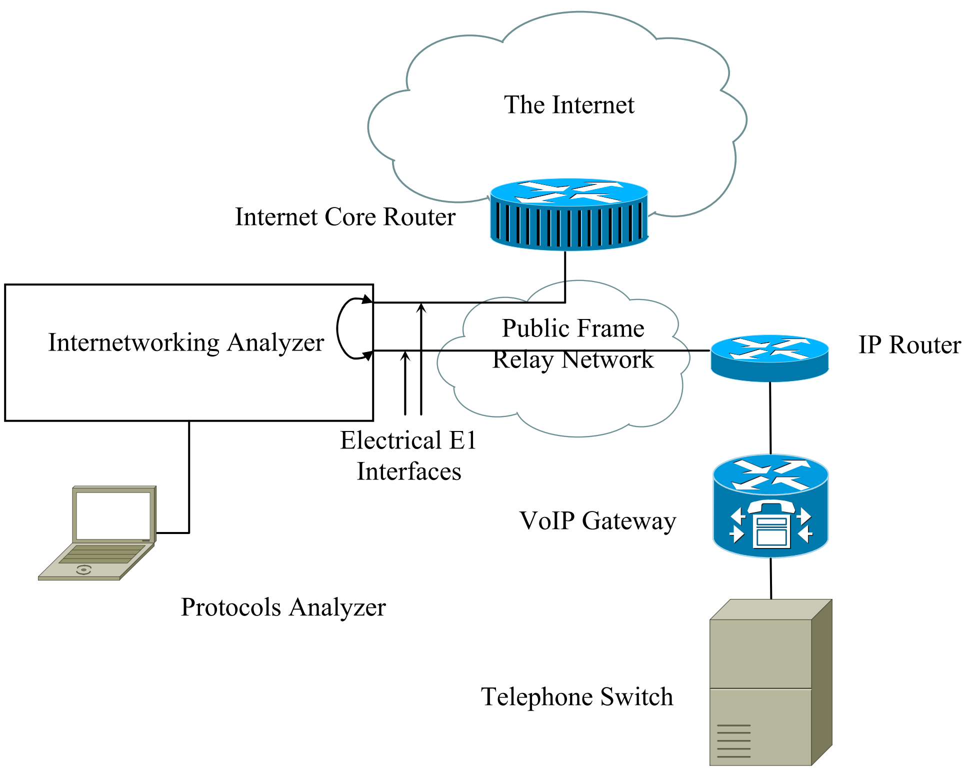


Figure (3.2.1): Connection of the Protocol Detector

3.2.2 Delay measurement

The round trip delay between the two end gateways was measured by using the standard ping command. The PING commands were issued from a computer at Sudatel Company network to the other gateway in USA. The computer was connected to the Internet core routers at the same hop distance as that of the VoIP gateway. The connection was made as shown in figure (3.2.2).

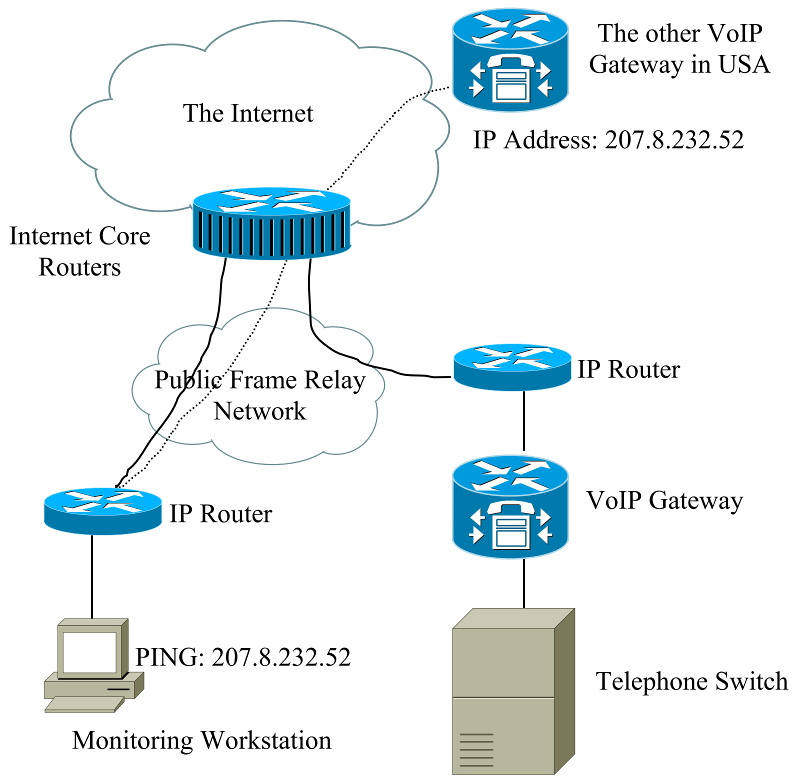


Figure (3.2.2): Arrangement of PING Command for Delay Test

3.2.3 Traffic load measurements

The traffic load statistics of the E1 trunk groups in both EZC and ITC solutions were gathered from Khartoum-South exchange call details records (CDR) by using an in-house developed traffic-analysis software tool (Sudatel Traffic Measurement Program, STM). The front-end monitor of this software, the back-end server and the exchange were connected as shown in Figure (3.2.3). The mediation module pulls the CDR file, which is in ASCII format, from the telephone switch and transfers it to the database table at the back-end server.

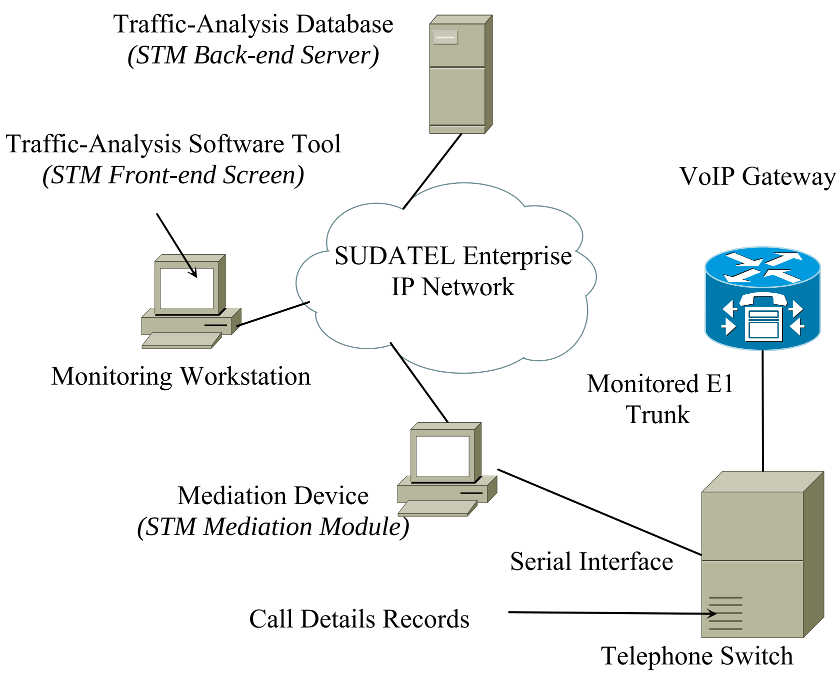


Figure (3.2.3): Connection of the Traffic Measurement Tool

3.2.4 Internet bandwidth utilization

The utilization statistics of the Internet link of EZC and ITC solutions were gathered from the corresponding ports at the Internet core routers by using a multi router traffic grapher (MRTG). The MRTG server and the monitoring workstation are connected as shown in Figure (3.2.4). The MRTG collects information about the transmitted and received volume of data at each port, and plots it in a daily, monthly and yearly basis.

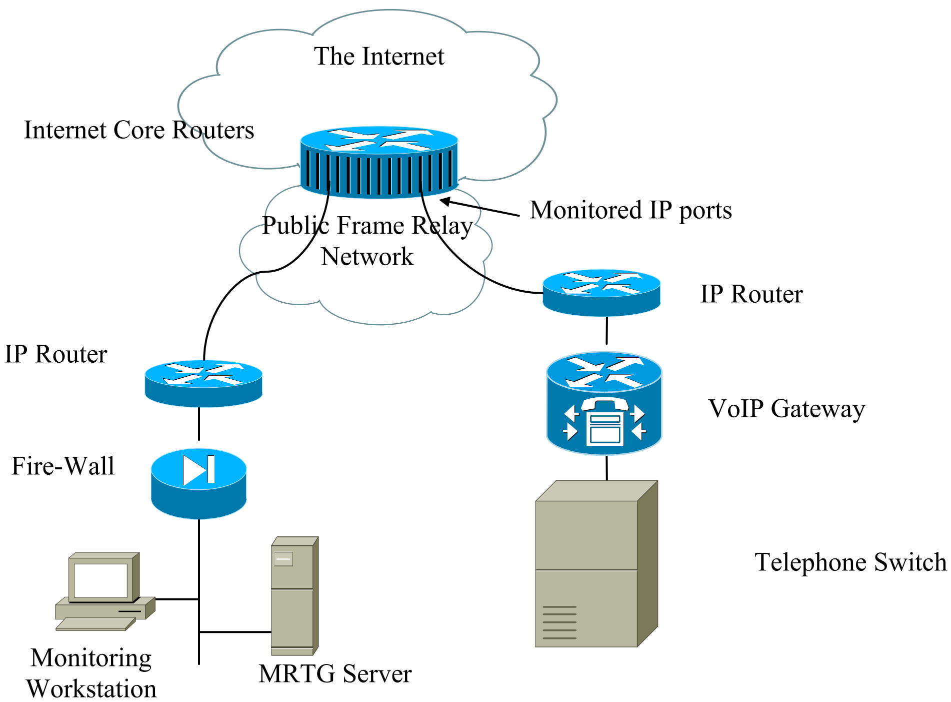


Figure (3.2.4): Connection of the Traffic Measurement Tool

4 RESULTS AND CALCULATIONS

4.1 Decoding of the protocols stack

A random sample of the transported packets by EZC gateway is captured and interpreted as listed in sample tables (4.1.1 – 4). The complete tables are contained in the attached CD disk. The captured packets are decoded and the protocols stack of most of the transported packets found to be constructed from the Frame Relay, IP, UDP, RTP, RTCP, and H.245 protocols.

Number	Delta Time	Interpretation
42214		DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DCE C/R=0 FECN=No Forward Explicit Congestion
42215	350 us	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DTE C/R=0 FECN=No Forward Explicit Congestion
42216	2.5 ms	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DCE C/R=0 FECN=No Forward Explicit Congestion
42217	770 us	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DTE C/R=0 FECN=No Forward Explicit Congestion
42218	2.1 ms	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DCE C/R=0 FECN=No Forward Explicit Congestion
42219	740 us	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DTE C/R=0 FECN=No Forward Explicit Congestion
42220	2.1 ms	DLCI=30 InformationType=Non D channel user information ControlField=0x03 M_bits=Modifier Function Bits S=DCE C/R=0 FECN=No Forward Explicit Congestion

Table (4.1.1): Packets Decoding, Frame Relay Protocol

Number	Delta Time	Destination	Source	Interpretation
42214		195.219.107.122	207.8.232.52	D=195.219.107.122 S=207.8.232.52 Protocol=UDP ID=20508 FragmentOffset=0 Total_Len=84
42215	350 us	207.8.232.52	195.219.107.122	D=207.8.232.52 S=195.219.107.122 Protocol=UDP ID=9078 FragmentOffset=0 Total_Len=84
42216	2.5 ms	195.219.107.122	207.8.232.52	D=195.219.107.122 S=207.8.232.52 Protocol=UDP ID=20509 FragmentOffset=0 Total_Len=84
42217	770 us	207.8.232.52	195.219.107.122	D=207.8.232.52 S=195.219.107.122 Protocol=UDP ID=9079 FragmentOffset=0 Total_Len=84
42218	2.1 ms	195.219.107.122	207.8.232.52	D=195.219.107.122 S=207.8.232.52 Protocol=UDP ID=20510 FragmentOffset=0 Total_Len=84
42219	740 us	207.8.232.52	195.219.107.122	D=207.8.232.52 S=195.219.107.122 Protocol=UDP ID=9080 FragmentOffset=0 Total_Len=84
42220	2.1 ms	195.219.107.122	207.8.232.52	D=195.219.107.122 S=207.8.232.52 Protocol=UDP ID=20511 FragmentOffset=0 Total_Len=84

Table (4.1.2): Packets Decoding, IP Protocol

Number	Delta Time	Interpretation
42214		D=0x0688 S=1672 Checksum Good
42215	350 us	D=0x0688 S=1672 Checksum Good
42216	2.5 ms	D=0x0688 S=1672 Checksum Good
42217	770 us	D=0x0688 S=1672 Checksum Good
42218	2.1 ms	D=0x0688 S=1672 Checksum Good
42219	740 us	D=0x0688 S=1672 Checksum Good

Table (4.1.3): Packets Decoding, UDP Protocol

Number	Delta Time	Interpretation
42214		FRAME TOO SHORT Flags=0x0E0E Ver=VAT Timestamp=55425.9902 Second(s) SSRC=2437480063 CSRC_Identifier=332956157
42215	350 us	FRAME TOO SHORT Flags=0x0E0E Ver=VAT Timestamp=48236.0990 Second(s) SSRC=3512859572 CSRC_Identifier=3321968741
42216	2.5 ms	Flags=0x0505 Ver=VAT Timestamp=55456.1078 Second(s) SSRC=73399457 CSRC_Identifier=876686365
42217	770 us	Flags=0x0909 Ver=VAT Timestamp=8580.6050 Second(s) SSRC=1117089345 CSRC_Identifier=672719273
42218	2.1 ms	Flags=0x0707 Ver=VAT Timestamp=27595.0825 Second(s) SSRC=333569091 CSRC_Identifier=3133237803
42219	740 us	Flags=0x0606 Ver=VAT Timestamp=43294.2656 Second(s) SSRC=480422087 CSRC_Identifier=2890799279

Table (4.1.4): Packets Decoding, RTP Protocol

Number	Delta Time	Interpretation
42214		FRAME TOO SHORT 0x0E Ver=0x00 Len=0x0E00
42215	350 us	FRAME TOO SHORT 0x0E Ver=0x00 Len=0x0E3A
42216	2.5 ms	FRAME TOO SHORT 0x05 Ver=0x00 Len=0x0543
42217	770 us	FRAME TOO SHORT 0x09 Ver=0x00 Len=0x096C
42218	2.1 ms	FRAME TOO SHORT 0x07 Ver=0x00 Len=0x070A
42219	740 us	FRAME TOO SHORT 0x06 Ver=0x00 Len=0x0675

Table (4.1.5): Packets Decoding, RTCP Protocol

Number	Delta Time	Destination	Source	Interpretation
42214		195.219.107.122	207.8.232.52	Frame Fragmented Command Msg=End Session Command Type=Non-Standard
42215	350 us	207.8.232.52	195.219.107.122	Indication Msg=Request Mode Release
42216	2.5 ms	195.219.107.122	207.8.232.52	Command Msg=Unknown
42217	770 us	207.8.232.52	195.219.107.122	Indication Msg=Request Multiplex Entry Release
42218	2.1 ms	195.219.107.122	207.8.232.52	Response Msg=Unknown
42219	740 us	207.8.232.52	195.219.107.122	Response Msg=Unknown

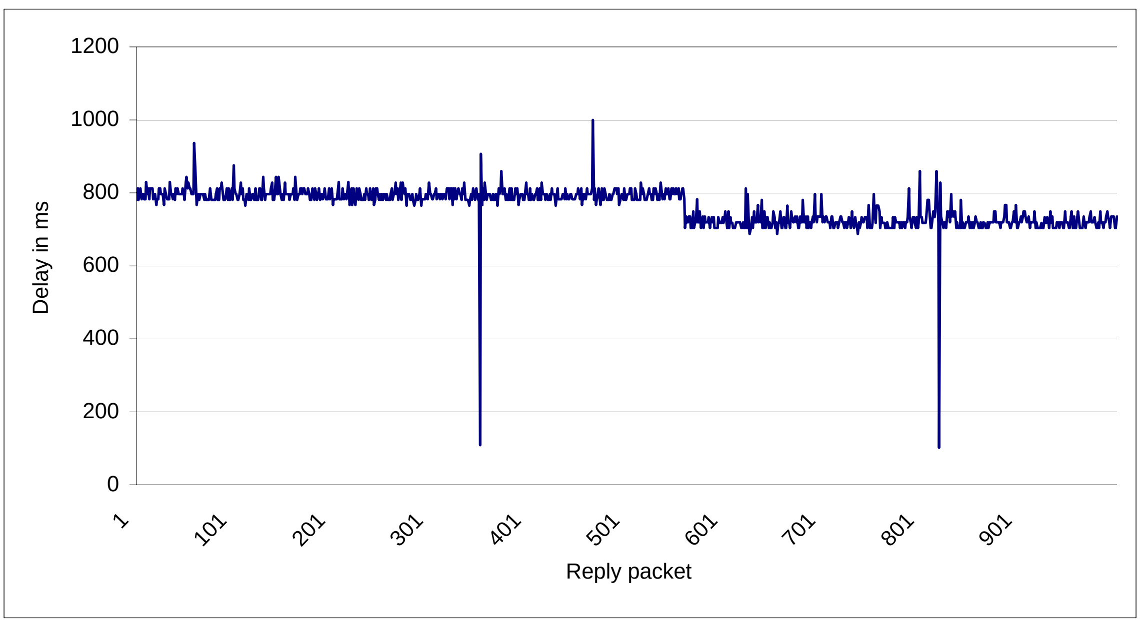
Table (4.1.6): Packets Decoding, H.225 Protocol

4.2 Measurement of the delays

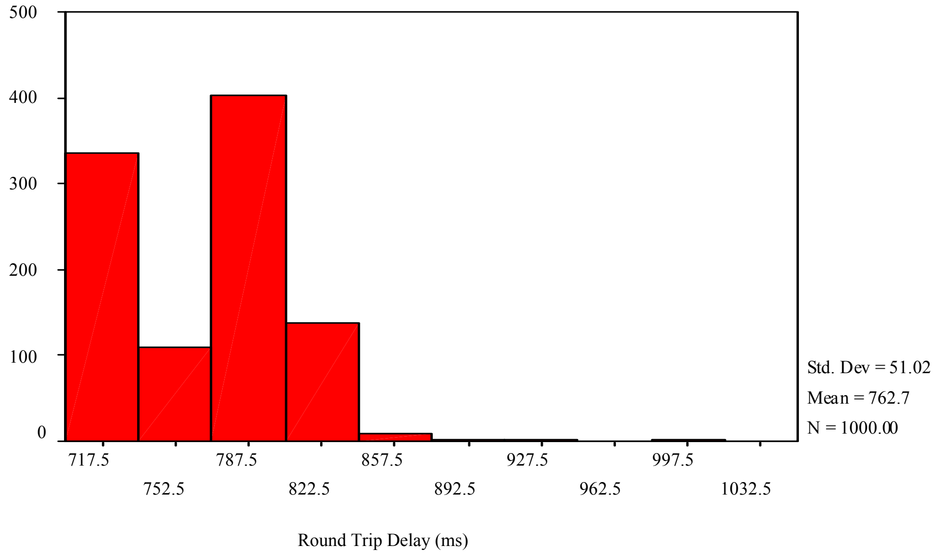
The round-trip delay between the two gateways of EZC are measured under a high traffic load, the readings are as listed in the sample table (4.2.1) and plotted in graphs (4.2.1) and (4.2.2). The average round trip delay is found to be 764.3 ms, which implies 382.15 ms one-way delay. This delay is acceptable according to the ITU-T G.114 Recommendation.

Packet	PING Results	Delay (ms)	Jitter (ms)
1	Reply from 207.8.232.52: bytes=64 time=781ms TTL=238	781	0
2	Reply from 207.8.232.52: bytes=64 time=812ms TTL=238	812	31
3	Reply from 207.8.232.52: bytes=64 time=781ms TTL=238	781	-31
4	Reply from 207.8.232.52: bytes=64 time=797ms TTL=238	797	16
5	Reply from 207.8.232.52: bytes=64 time=813ms TTL=238	813	16
6	Reply from 207.8.232.52: bytes=64 time=782ms TTL=238	782	-31
7	Reply from 207.8.232.52: bytes=64 time=797ms TTL=238	797	15
8	Reply from 207.8.232.52: bytes=64 time=797ms TTL=238	797	0
9	Reply from 207.8.232.52: bytes=64 time=782ms TTL=238	782	-15
10	Reply from 207.8.232.52: bytes=64 time=782ms TTL=238	782	0

Table (4.2.1): PING Command Results



Graph (4.2.1): Round Trip Delay Between EZC VoIP Gatways



Graph (4.2.2): Histogram of Round Trip Delay Between EZC VoIP Gatways

4.3 Tele-traffic engineering characteristics

Traffic load statistics of EZC and ITC E1 voice trunks were measured during the period from 31st March 2002 to 1st April 2003. Samples of the measurement results are listed in tables (4.3.1) and (4.3.2) respectively. The rest of the data is contained in the attached CD disk. The results are analyzed using statistics analytical tools (*SPSS R 10.0.5 and Microsoft Excel 2000*).

Date	Number of Circuits (Circuit)	Seizure Calls (Call)	Seizure Occupancy (Erlang)	Answered Calls (Call)	Conversation Occupancy (Erlang)
01/04/2002 00:15	30	80	21.6	31	23.8
01/04/2002 00:30	30	87	19.8	31	18.4
01/04/2002 00:45	30	54	21.2	17	13.8
01/04/2002 01:00	30	31	15.5	20	21.2
01/04/2002 01:15	30	30	10.5	14	12.8
01/04/2002 01:30	30	18	7.7	11	6.6
01/04/2002 01:45	30	22	7.5	3	8
01/04/2002 02:00	30	12	3.1	6	5.1
01/04/2002 02:15	30	9	1.5	2	3.8
01/04/2002 02:30	30	9	1.1	2	1.2
01/04/2002 02:45	30	6	2	1	0.8
01/04/2002 03:00	30	7	3.2	2	1
01/04/2002 03:15	30	8	4.1	6	4.8

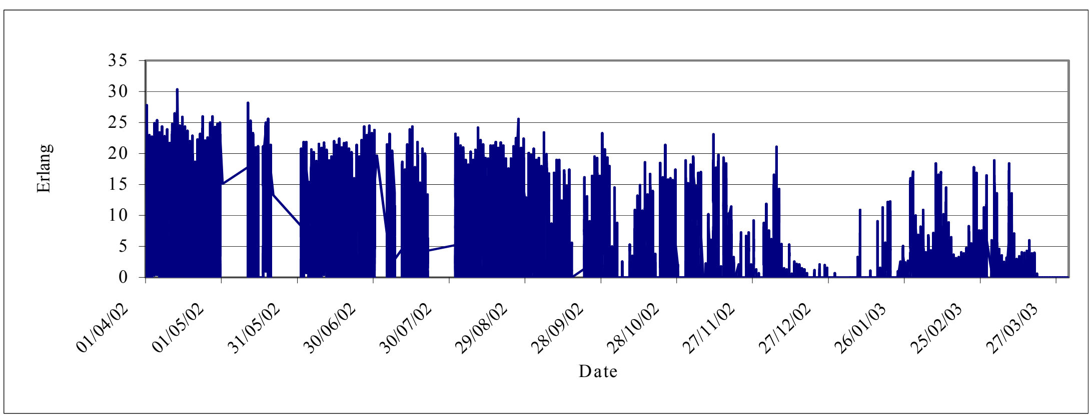
Table (4.3.1): Sample of traffic load statistics of EZC E1 voice trunk
(Measured during 31st March 2002 to 1st April 2003)

Date	Number of Circuits (Circuit)	Seizure Calls (Call)	Seizure Occupancy (Erlang)	Answered Calls (Call)	Conversation Occupancy (Erlang)
01/04/2002 00:15	30	76	5.9	43	6.4
01/04/2002 00:30	30	28	2.8	16	1.6
01/04/2002 00:45	30	2	2.1	2	3.1
01/04/2002 01:00	30	3	2	0	0
01/04/2002 01:15	30	1	1.5	2	3
01/04/2002 01:30	30	1	0.7	1	0.8
01/04/2002 01:45	30	2	0.3	1	0.7
01/04/2002 02:00	30	4	0	0	0
01/04/2002 02:15	30	1	0	0	0
01/04/2002 02:30	30	0	0	0	0
01/04/2002 02:45	30	0	0	0	0
01/04/2002 03:00	30	0	0	0	0
01/04/2002 03:15	30	4	0.1	1	0

Table (4.3.2): Sample of traffic load statistics of ITC E1 voice trunk
(Measured during 31st March 2002 to 1st April 2003)

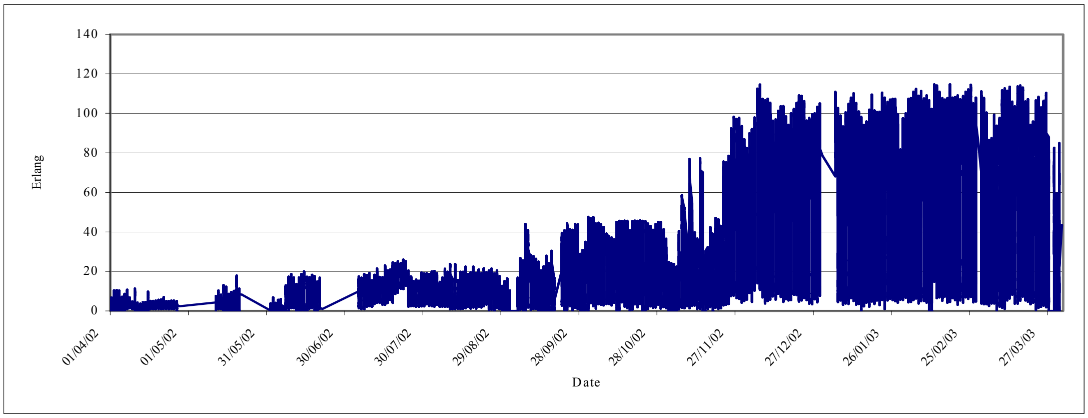
4.3.1 Measurement of the traffic load intensity

The traffic load intensity in Erlang is calculated from the measured occupancy of the trunk. The traffic intensity during the whole year for EZC and ITC trunks are plotted in graphs (4.3.1.1) and (4.3.1.2) respectively.



Graph (4.3.1.1): Traffic Load of EZC Trunk (in Erlang)

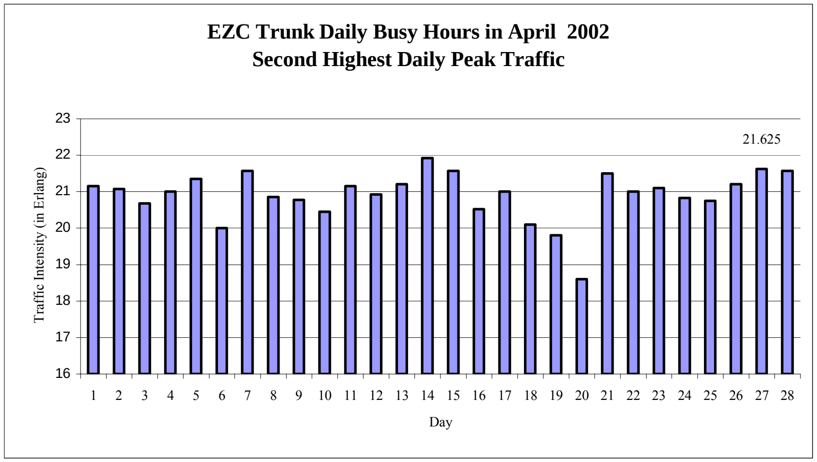
April 2002 - March 2003

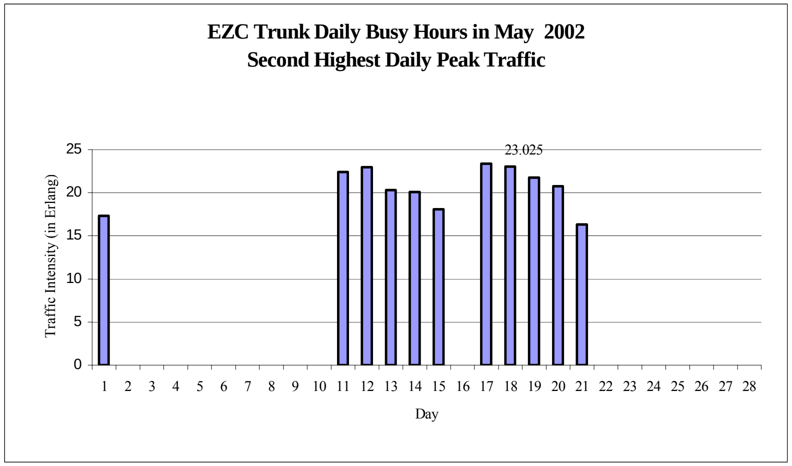


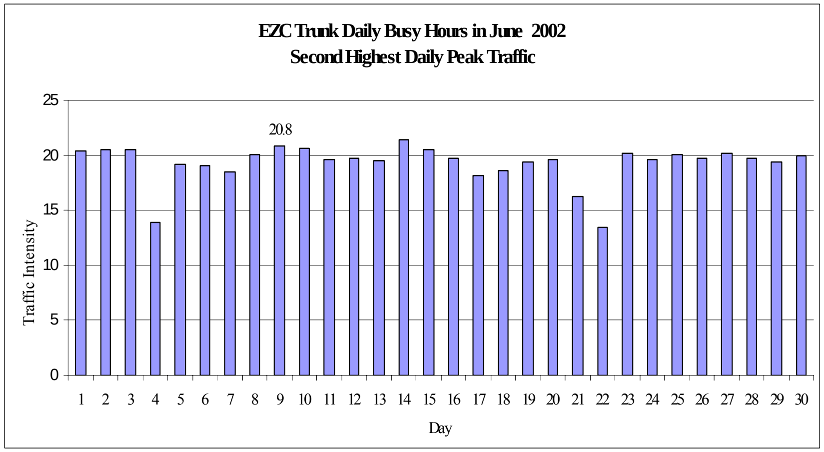
Graph (4.3.1.2): Traffic Load of ITC Trunk (in Erlang)
April 2002 - March 2003

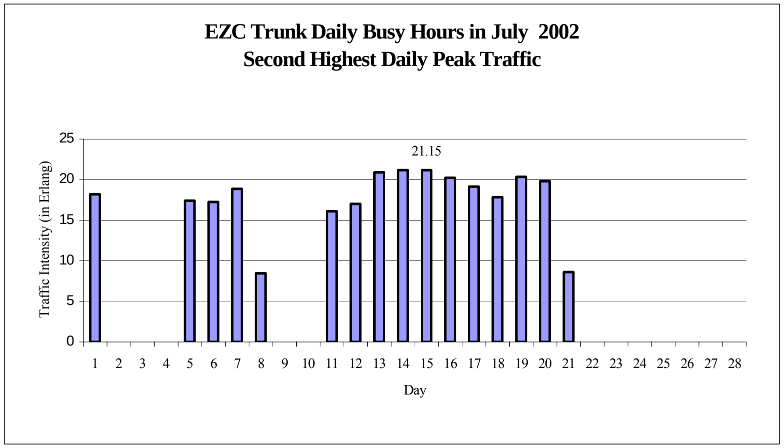
4.3.2 Yearly Representative Value of the high traffic load

The Yearly Representative Value (YRV) of EZC and ITC trunks are found as described in ITU-T E.500. The busy hour for every day in each month are found. Then the second highest busy hour in every month are selected as shown in graphs (4.3.2.1 - 3) and (4.3.2.5 - 7). Finally, the YRV is calculated from the second highest reading in the year as shown in graphs (4.3.2.4) and (4.3.2.8). The YRV of EZC and ITC trunks are found to be 21.45 Erlang and 112.14 Erlang respectively.



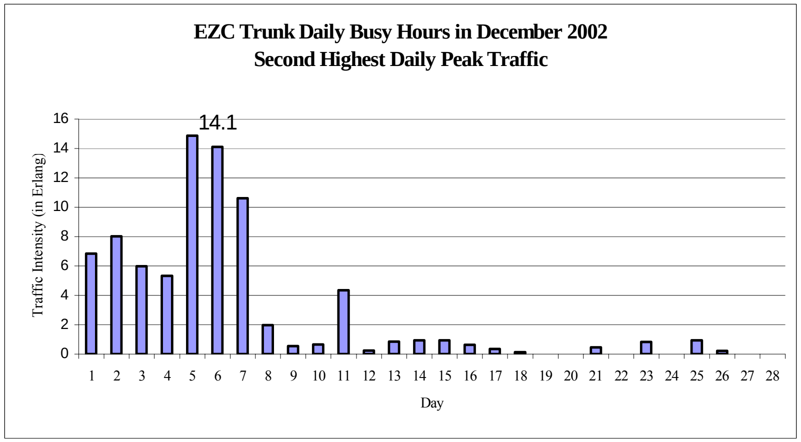


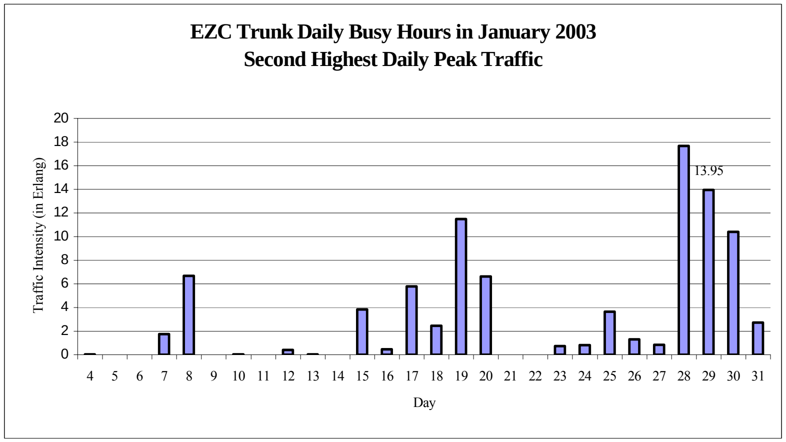


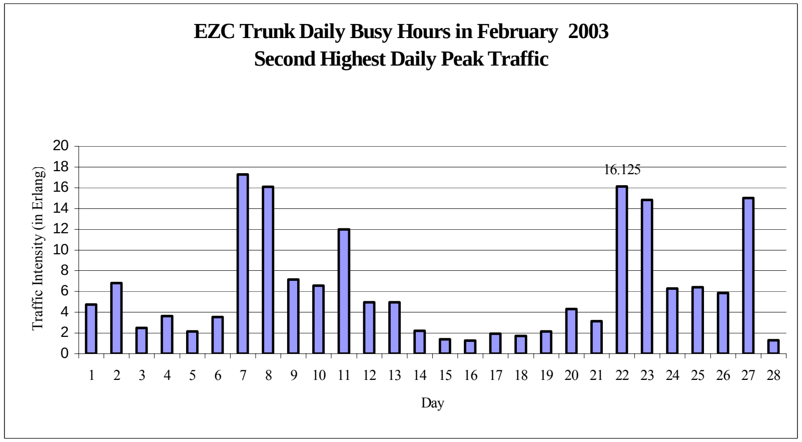


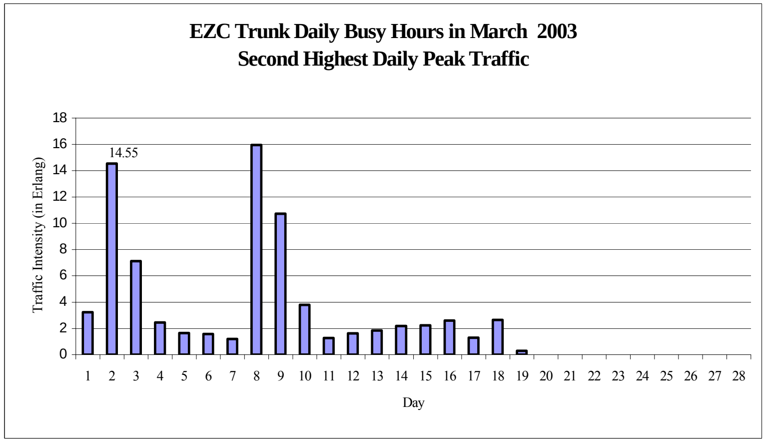
Graph (4.3.2.1): Daily Busy Hour for EZC Trunk

Graph (4.3.2.2): Daily Busy Hour for EZC Trunk

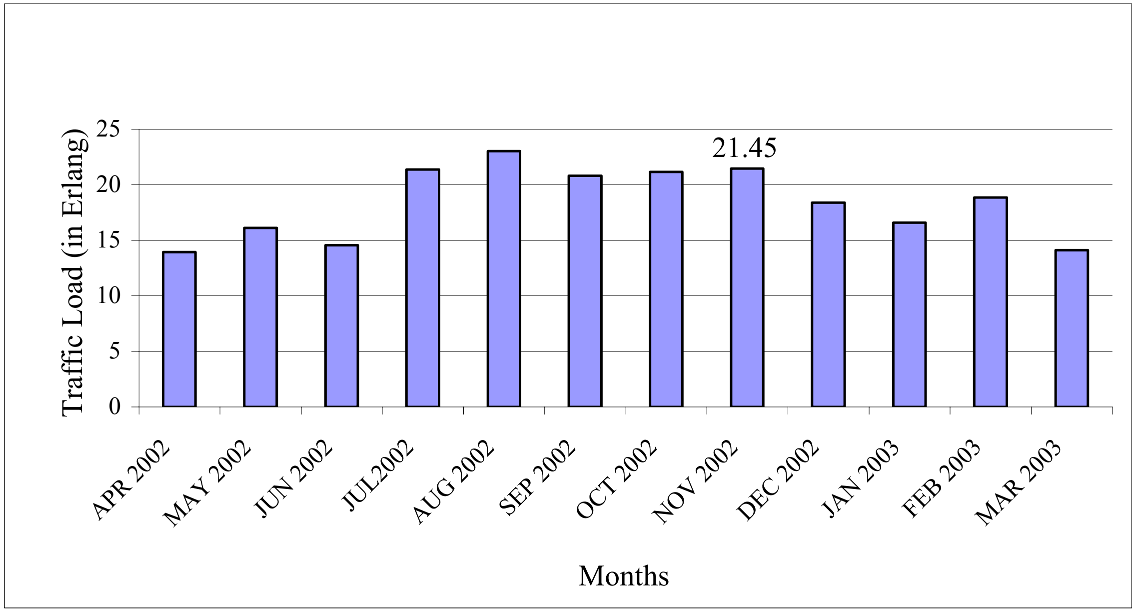






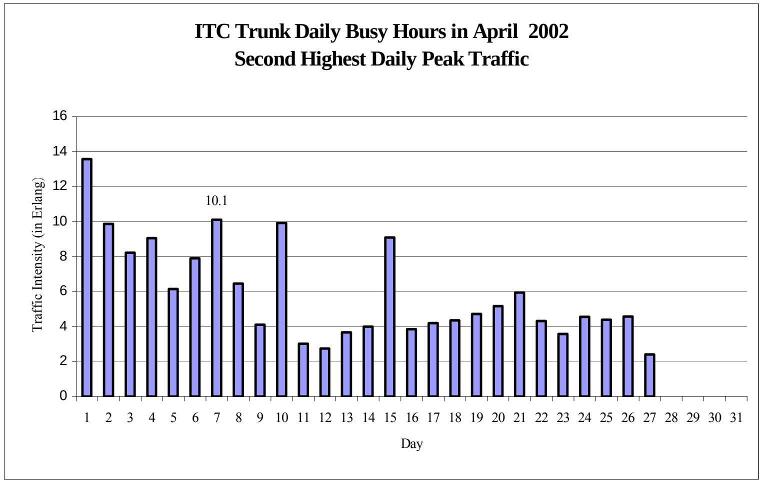


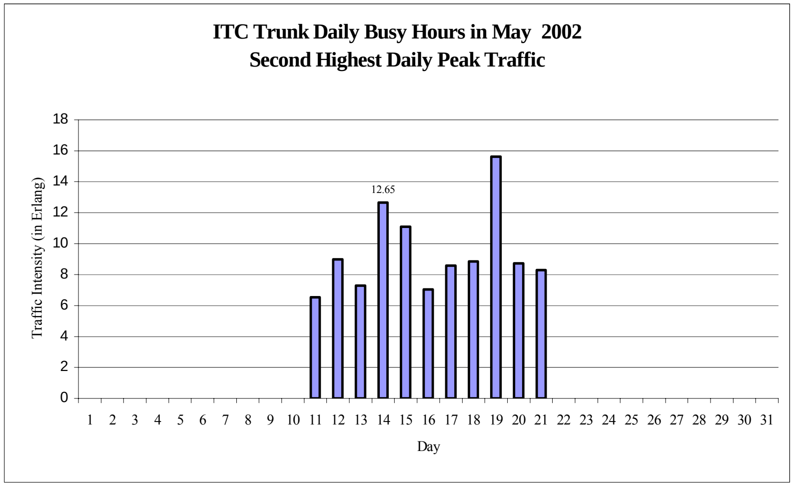
Graph (4.3.2.3): Daily Busy Hour for EZC Trunk

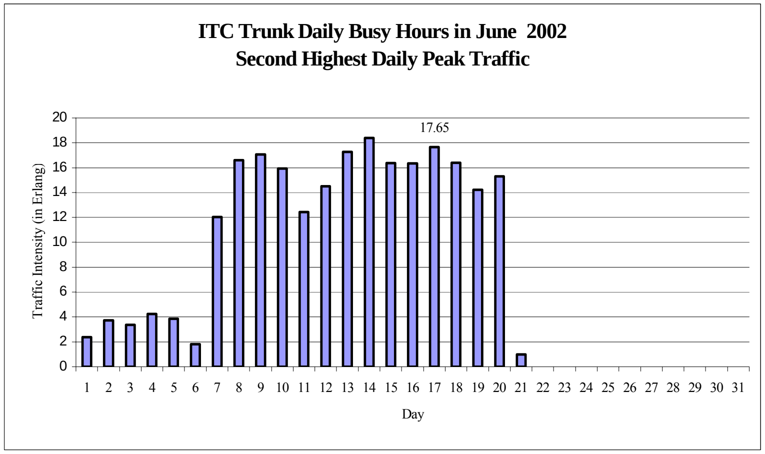


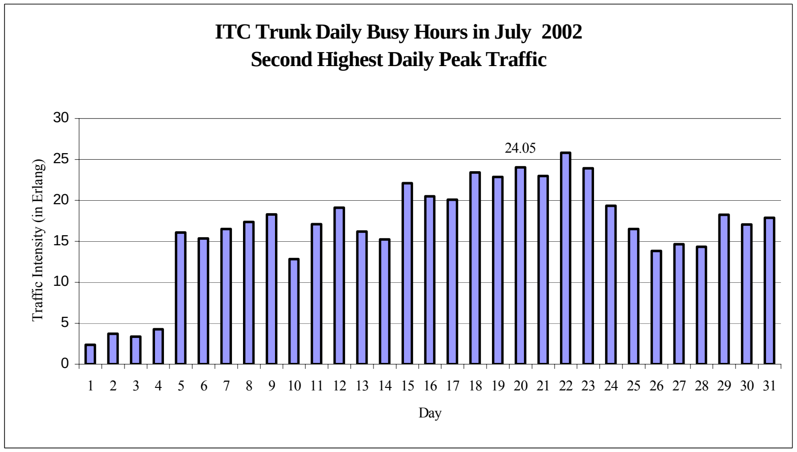
Graph (4.3.2.4): Monthly High Load Traffic Intensity of EZC Trunk

Yearly Representative Value (YRV)

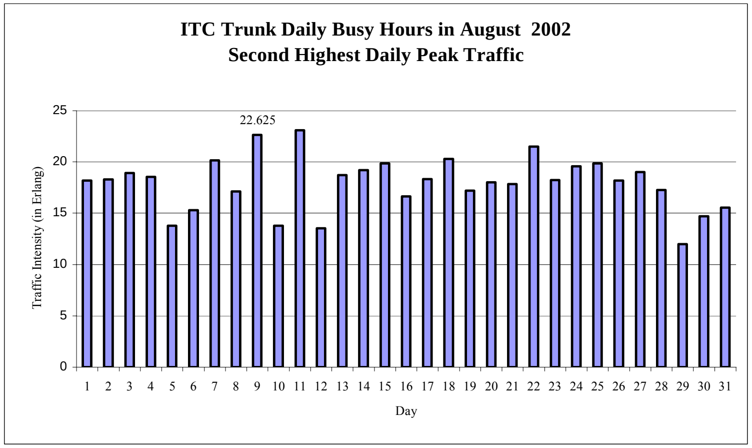


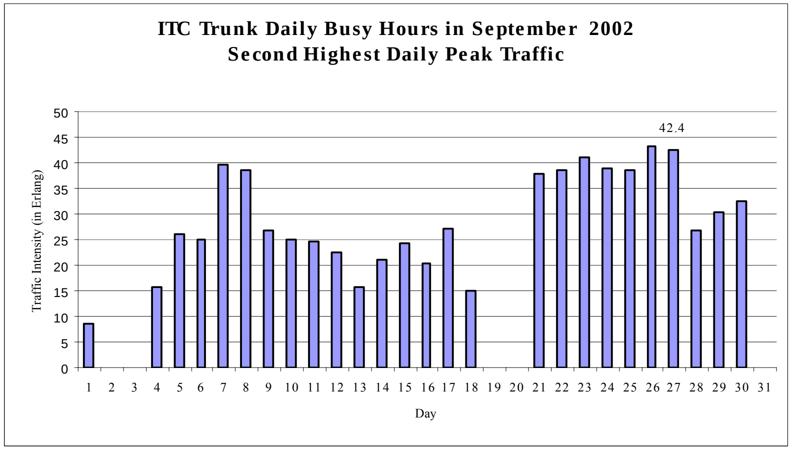


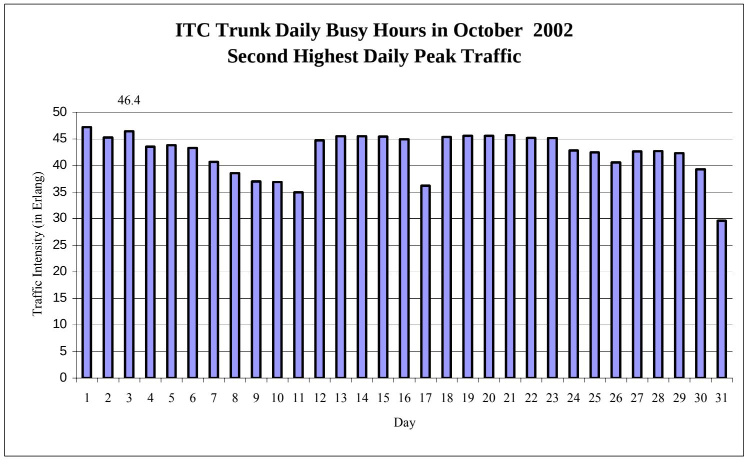


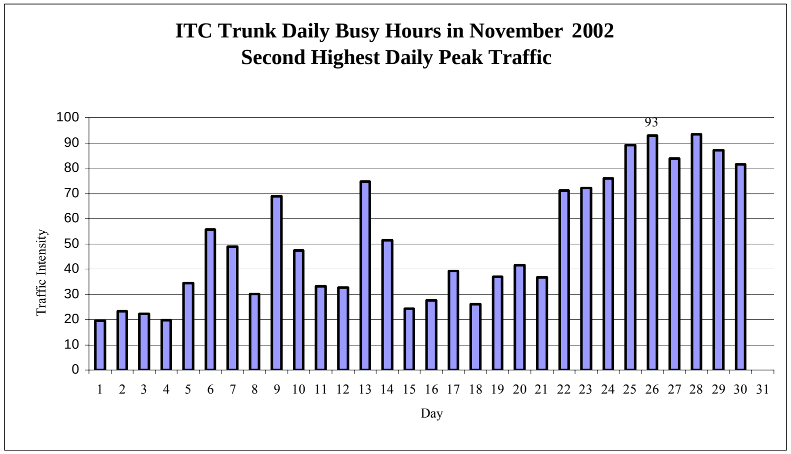


Graph (4.3.2.5): Daily Busy Hour for ITC Trunk

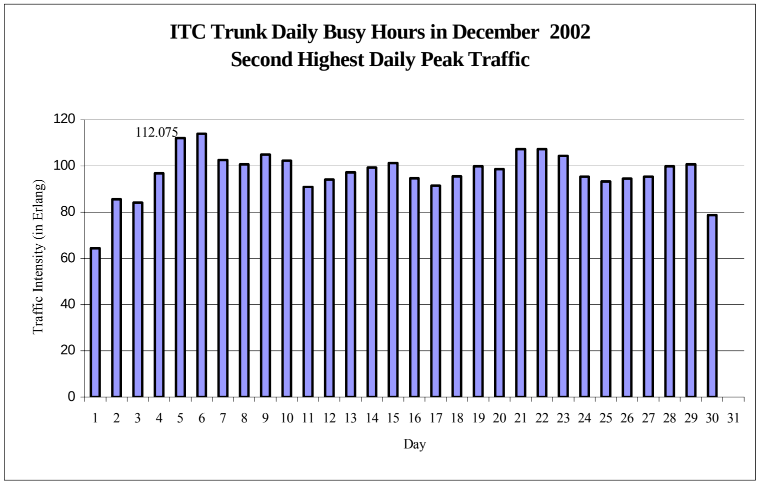


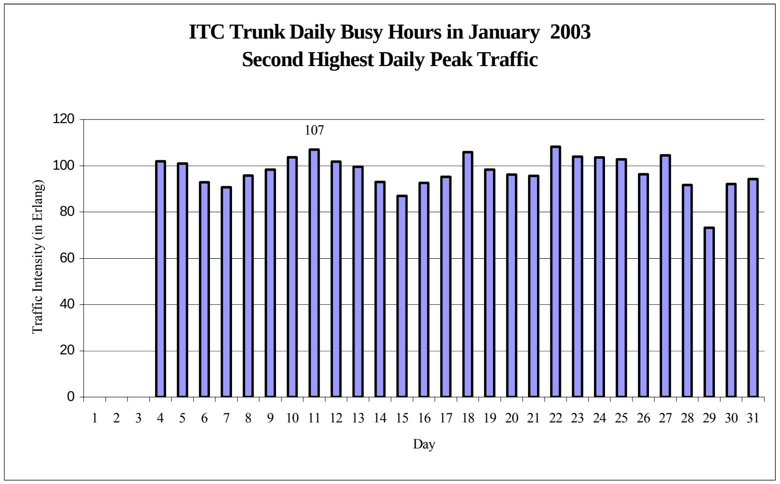


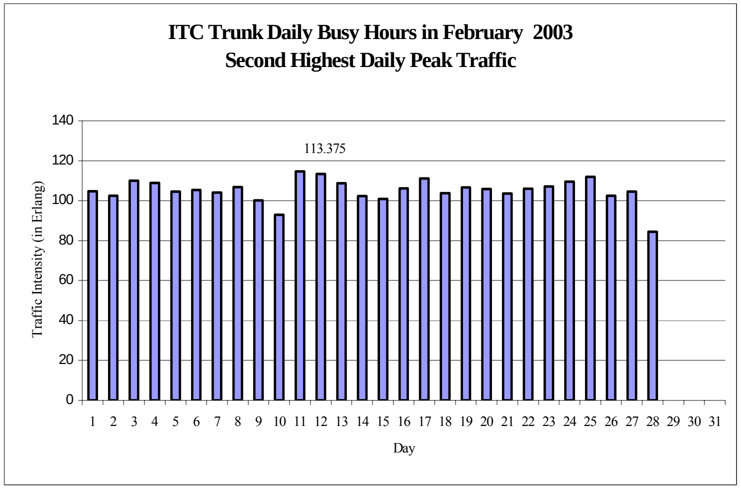


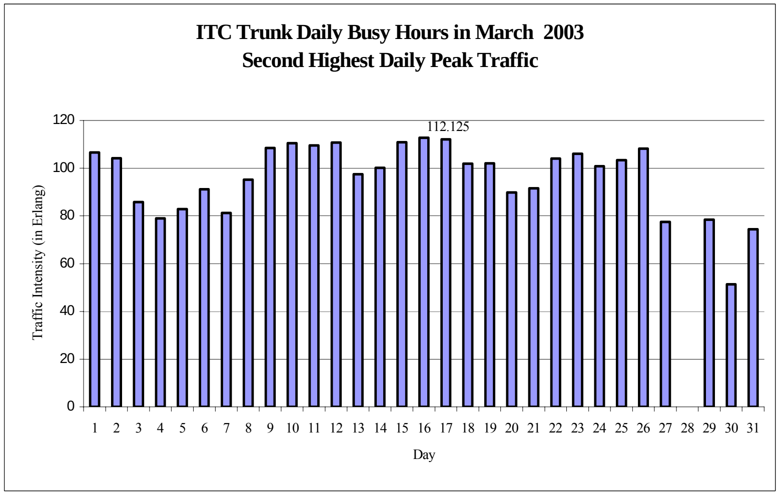


Graph (4.3.2.6): Daily Busy Hour for ITC Trunk

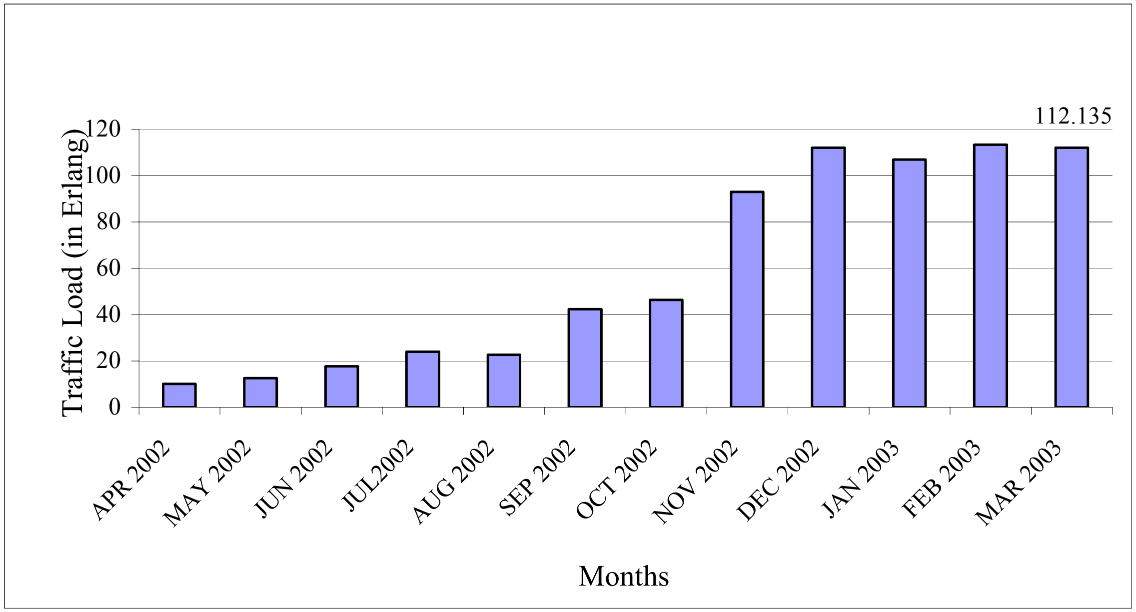








Graph (4.3.2.7): Daily Busy Hour for ITC Trunk

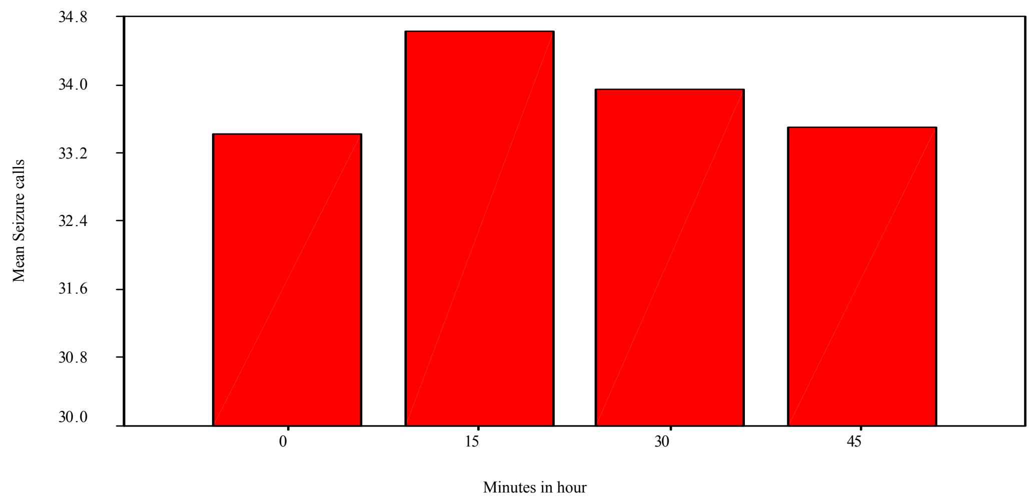


Graph (4.3.2.8): Monthly High Load Traffic Intensity of ITC Trunk

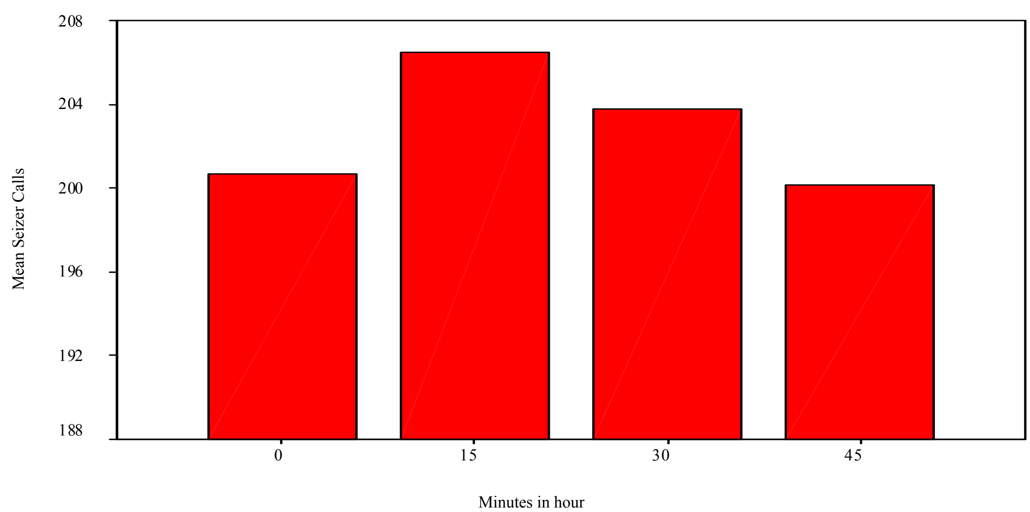
Yearly Representative Value (YRV)

4.3.3 Calls arrival pattern

The mean of the call numbers arrived per hour of EZC and ITC trunks are found to be as plotted in graphs (4.3.3.1) and (4.3.3.2) respectively. It can be observed that the call arrival pattern is neither peaked nor smooth.



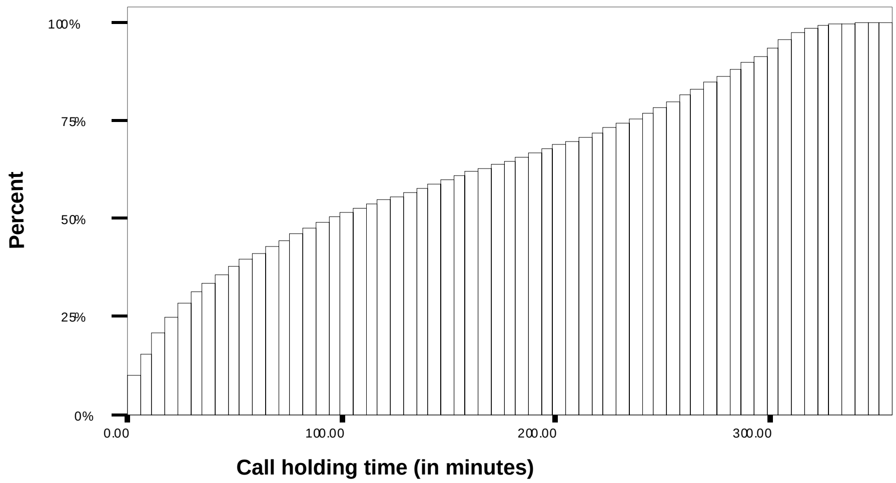
Graph (4.3.3.1): Pattern of Calls Arrival per Hour of EZC Trunk



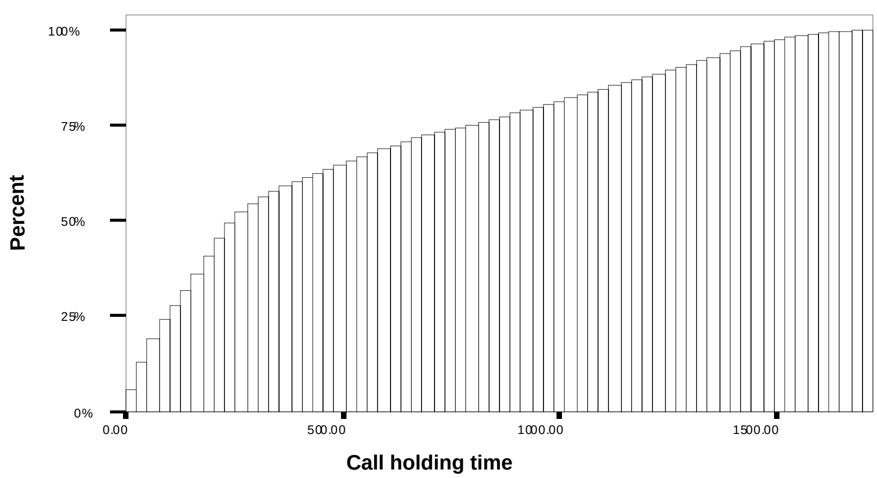
Graph (4.3.3.2): Pattern of Calls Arrival per Hour of ITC Trunk

4.3.4 Holding times pattern

The histograms of the holding time are obtained from the observed values by using SPSS as shown in graphs (4.3.4.1) and (4.3.4.2). The histogram seems to have an exponential distribution function.



Graph (4.3.4.1): Histogram of Call Holding Time of EZC Trunk



Graph (4.3.4.2): Histogram of Call Holding Time of ITC Trunk

The theoretical exponential distribution function F(x) of a random value X for n countable values x_n is given by:

$$F(x) = 1 - e^{-\lambda x} \qquad\qquad \lambda > 0, x \geq 0$$

Where λ, for this function, is obtained from the expectation value as:

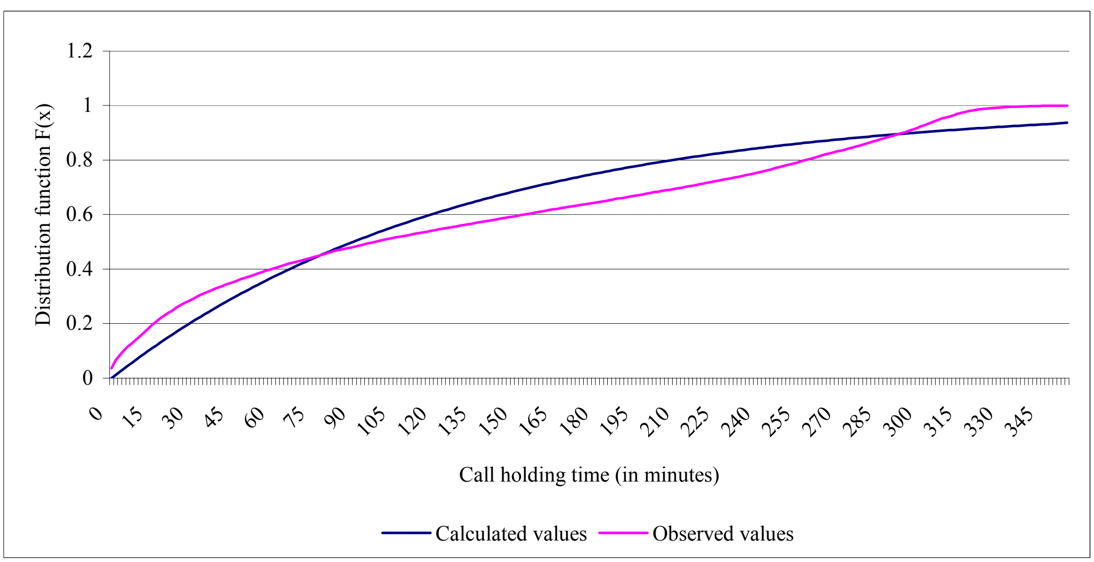
$$\frac{1}{\lambda} = \mathrm{E}(x) = \sum_{j=1}^{j=n} x_j * P\{X = x_j\}$$

Therefore λ is equal,

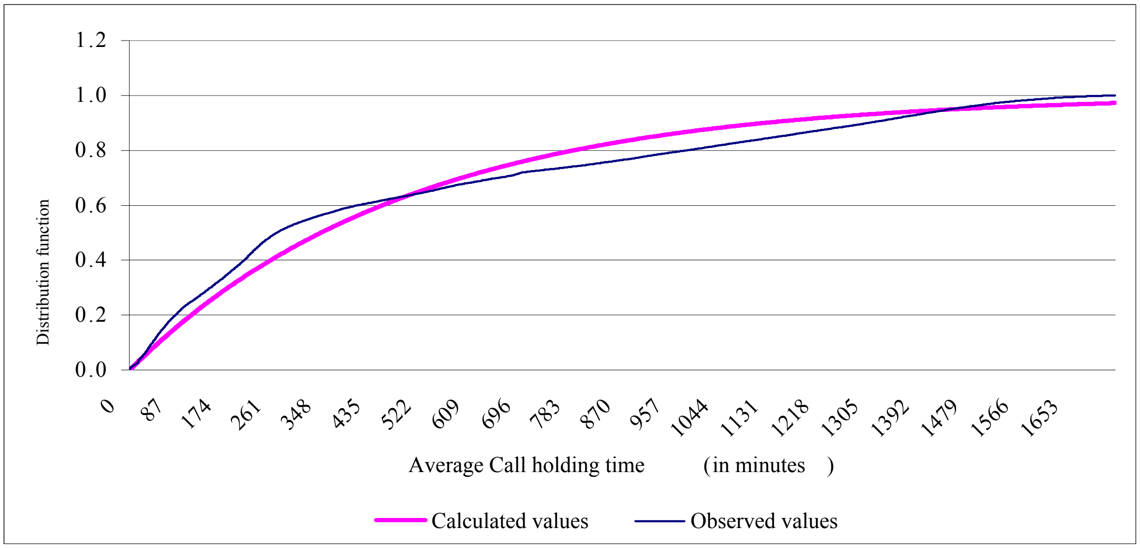
for EZC trunk $\lambda = 0.0077455734$

for ITC trunk $\lambda = 0.0020828868$

By substituting the theoretical value of λ, the function F(x) for each trunk is found as plotted in graphs (4.3.4.3) and (4.3.4.4). Statistical tests are carried out to compare between functions obtained from the observed values and functions obtained from theory. Tests verify that the holding time patterns have exponential distribution functions.



Graph (4.3.4.3): Distribution Function of EZC Holding Time



Graph (4.3.4.4): Distribution Function of ITC Holding Time

4.3.5 Modeling the traffic load

The calls that arrive and find these trunks busy are rerouted, as there are alternative trunks thus eliminating farther trials of busy resources. Therefore it can be assumed that the blocked calls are always cleared. The incoming calls in this traffic model can be assumed generated by an infinite number of sources.

With these assumptions and with the results obtained in 4.3.3 and 4.3.4 above, the traffic models of EZC and ITC voice trunks can be considered as Erlang-B models.

4.3.6 Evaluation of the trunks capacities

To evaluate the design values of these trunks, first the voice leg in each solution is considered. The blockage is assumed to be not more than one percentage (P.01), and then the numbers of PSTN circuits required for the measured traffic load above are obtained by using the table of Erlang formula in ITU-T E.800 Recommendation. It is found to be equal to 32 circuits for EZC trunk and 130 circuits for ITC trunk. Comparing these values with the used ones, it is clear that the numbers of circuits in the implemented trunks are the nearest value to the practical number of E1 trunks that suits this volume of traffic, but they are slightly less than what have to be in the ideal design, by 6.25% and 7.7%, respectively.

For the IP legs, providing that G.729 coding technique, compressed RTP header and voice detection activity are used, the IP bandwidth needed for EZC is:

$$=(9.6Kbps)(30\text{circuits}) = 288Kbps$$

The IP bandwidth needed for ITC is:

$$=(9.6Kbps)(120\text{circuits}) = 1152Kbps$$

Comparing these values with the used ones, it is clear that the implemented bandwidths are the nearest values that can be ordered commercially from the local service provider, but EZC bandwidth is slightly less (11.11%) than what have to be in the recommended design while ITC bandwidth is (142.22%) greater. ITC is expanding its transport requirement to meat the growing demand.

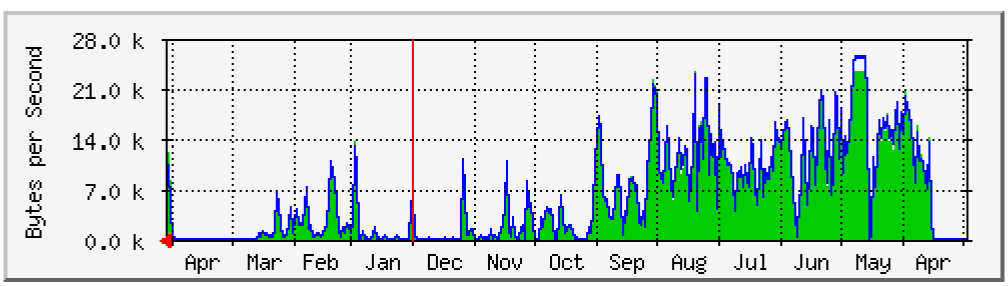
4.4 Calculation of Internet bandwidth utilization

The Internet link utilization for EZC and ITC bandwidths are measured and plotted by the MRTG as shown in graphs (4.4.1) and (4.4.2) respectively. The actual peaks reached are:

189.6 Kbps (74%)	in EZC link
436.8 Kbps (85.3%)	in ITC link

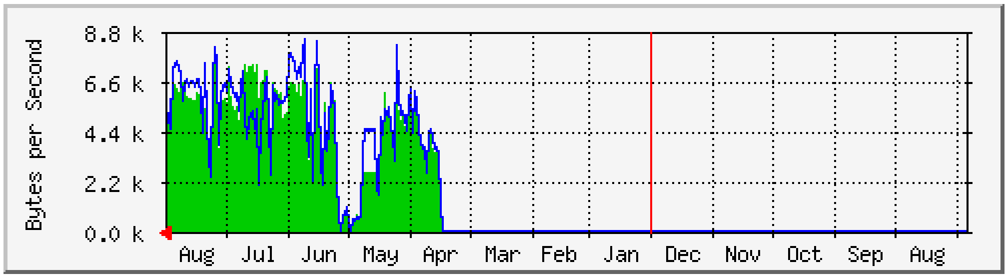
The average usages of the links are:

50.2 Kbps (19.6%)	in EZC link
240.8 Kbps (47%)	in ITC link



Max In: 23.7 kB/s (74.0%)	Average In: 6152.0 B/s (19.2%)	Current In: 19.1 kB/s (59.6%)
Max Out: 25.5 kB/s (79.8%)	Average Out: 6161.0 B/s (19.3%)	Current Out: 9980.0 B/s (31.2%)

Graph (4.4.1): Snap Shot of MRTG for EZC Port (Apr 2002 – Apr 2003)



Max In: 7502.0 B/s (23.4%) Average In: 4856.0 B/s (15.2%) Current In: 7237.0 B/s (22.6%)
Max Out: 8509.0 B/s (26.6%) Average Out: 4985.0 B/s (15.6%) Current Out: 5537.0 B/s (17.3%)

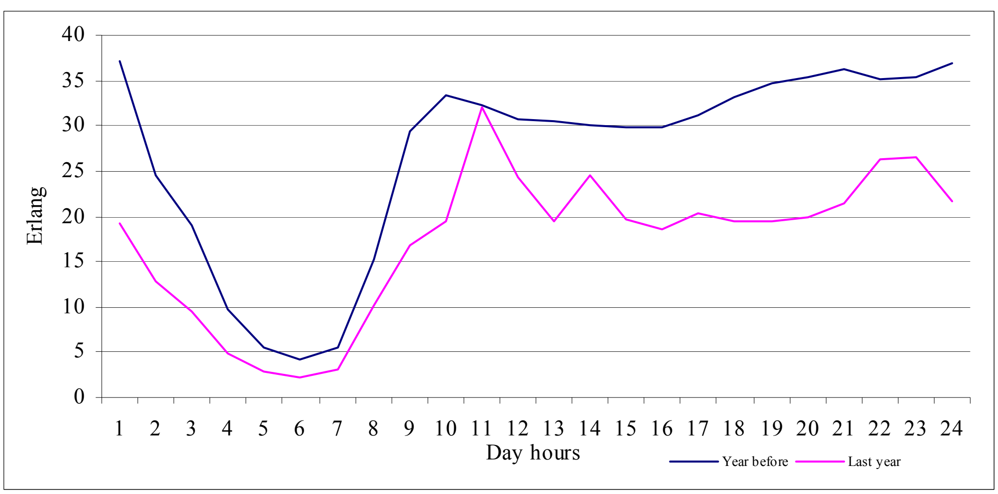
Graph (4.4.1): Snap Shot of MRTG for ITC Port (Apr 2002 – Aug 2002)

4.5 Traffic statistics

The traffic statistics of EZC and ITC trunks are compared with that of ATT&T ordinary voice trunk.

4.5.1 Effects of VoIP over ordinary trunk

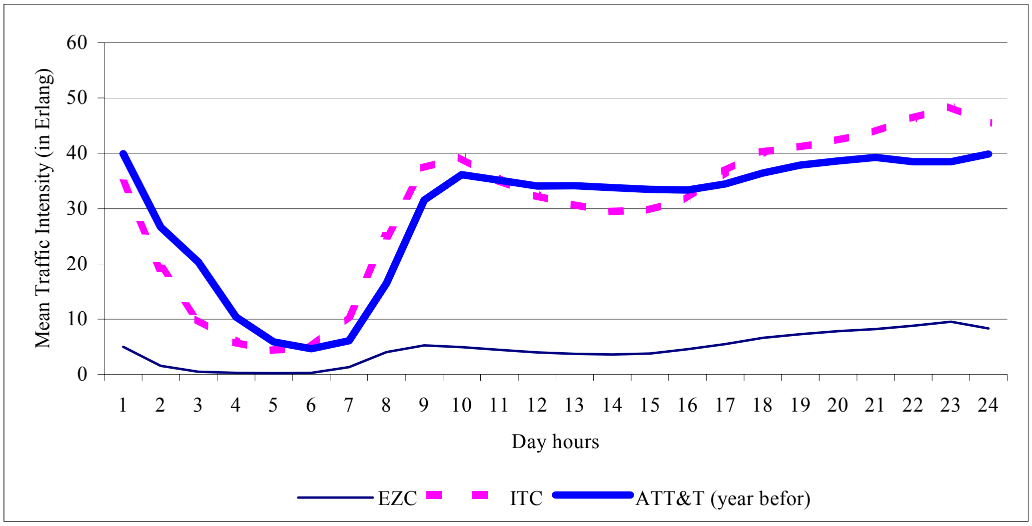
The traffic load of AT&T ordinary trunk for the year before and the year after introducing EZC and ITC VoIP trunks are compared to each other as shown in graph (4.5.1). The traffic carried has dropped down after introducing VoIP trunk with a ratio of more than 38% in average.



Graph (4.5.1): Mean Traffic Carried by ATT&T Trunks
Comparing The Year before and The Year after Introducing VoIP Trunks

4.5.2 Daily traffic profile

The daily traffic profile for EZC and ITC trunks are compared with that of ATT&T trunk of the year before introducing VoIP Trunks. Profiles show that the traffic pattern during the day is the same for all trunks. The high traffic is during the first hour at early morning, and it also during the period from 09:00 at morning to 24 at evening, as plotted in graph (4.5.2).

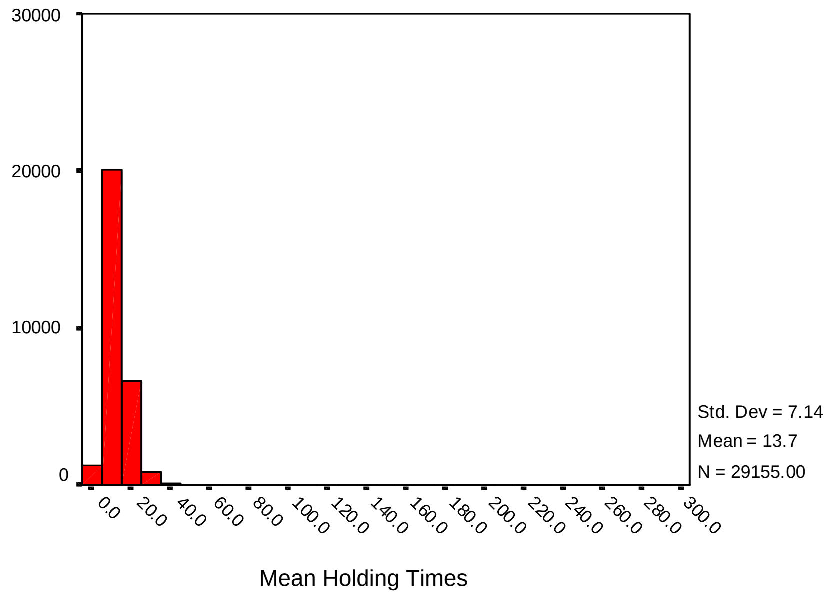


Graph (4.5.2): Traffic Intensity Daily Profile

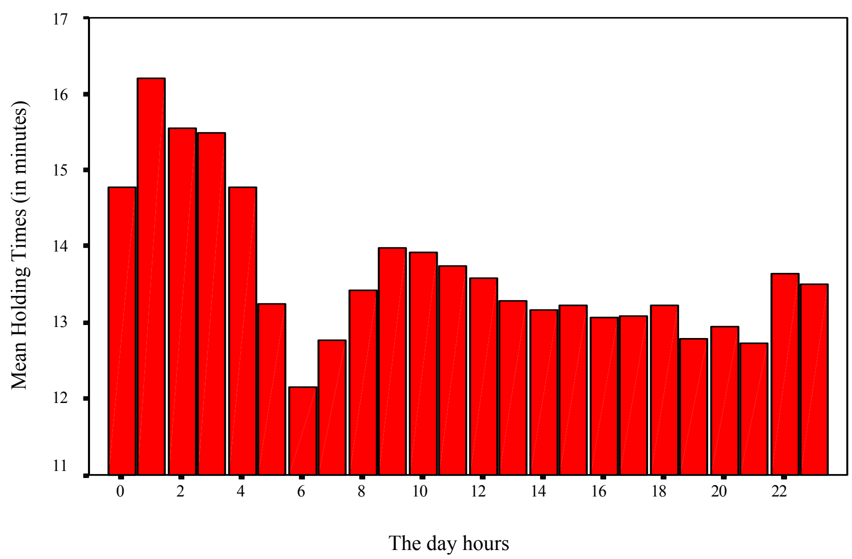
Comparison between EZC, ITC and ATT&T Trunks

4.5.3 Mean call holding time

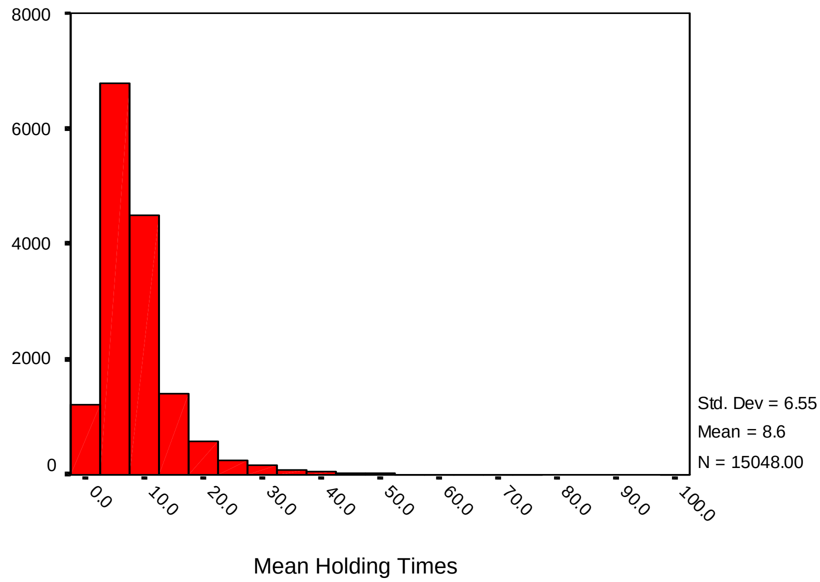
The average holding times for all trunks are obtained as shown in graphs (4.5.3.1) to (4.5.3.6). EZC has a mean holding time of 8.6 minutes, ITC has 13.7 minutes, and ATT&T has 8.4 minutes. The VoIP trunks show, relatively, a higher call holding time.



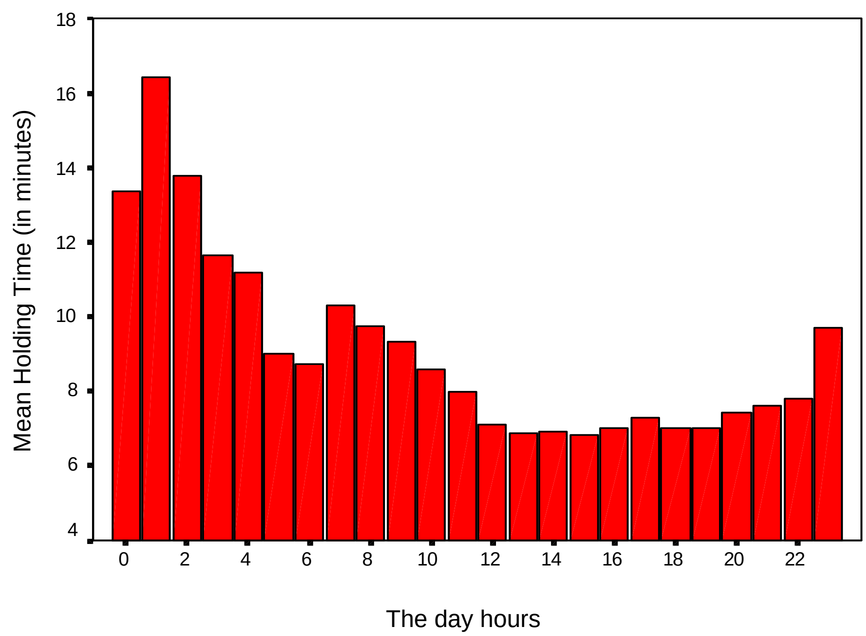
Graph (4.5.3.1): Histogram of The Mean Holding Time of ITC Trunk



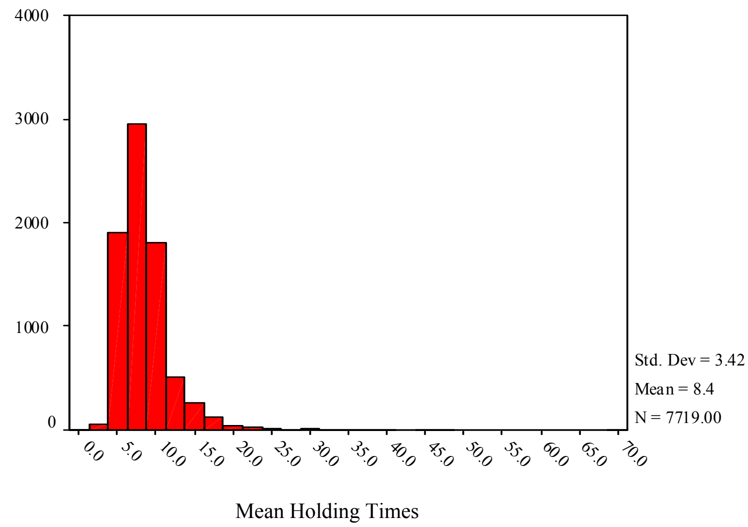
Graph (4.5.3.2): Holding Time of ITC Trunk per Day Hours for 12 Months



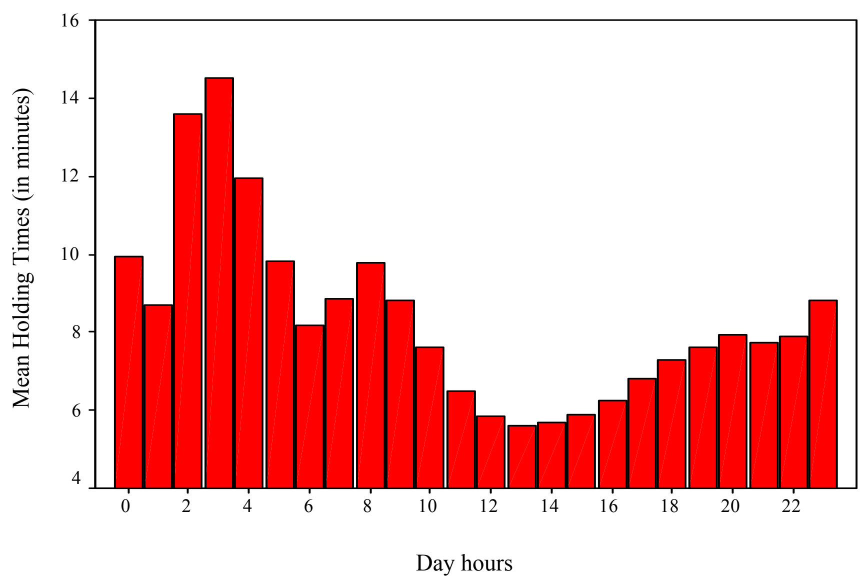
Graph (4.5.3.3): Histogram of The Mean Holding Time of EZC Trunk



Graph (4.5.3.4): Mean Holding Time of EZC Trunk per Day Hours for 12 Months



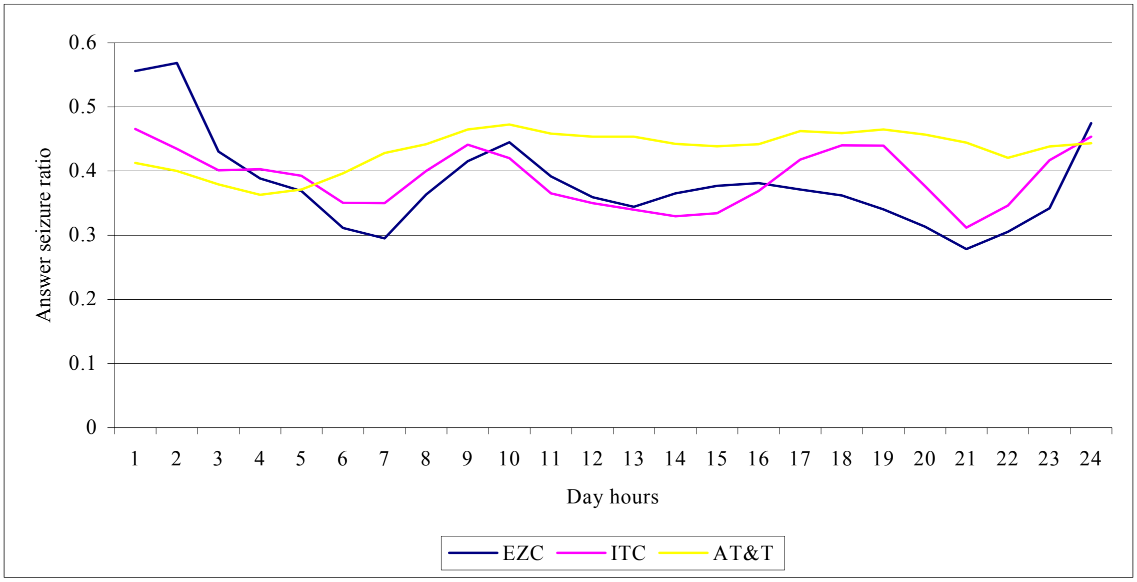
Graph (4.5.3.5): Histogram of The Mean Holding Time of AT&T Trunk



Graph (4.5.3.6): Mean Holding Time of AT&T Trunk per Day Hours for 12 Months

4.5.4 Answer seizure ratio (ASR)

The answer to seizure ratio (ASR) of EZC, ITC and AT&T trunks are compared as shown in graph (4.5.4). Mean ASR of EZC is 0.38, ITC is 0.39 and AT&T is 0.43, which is higher.



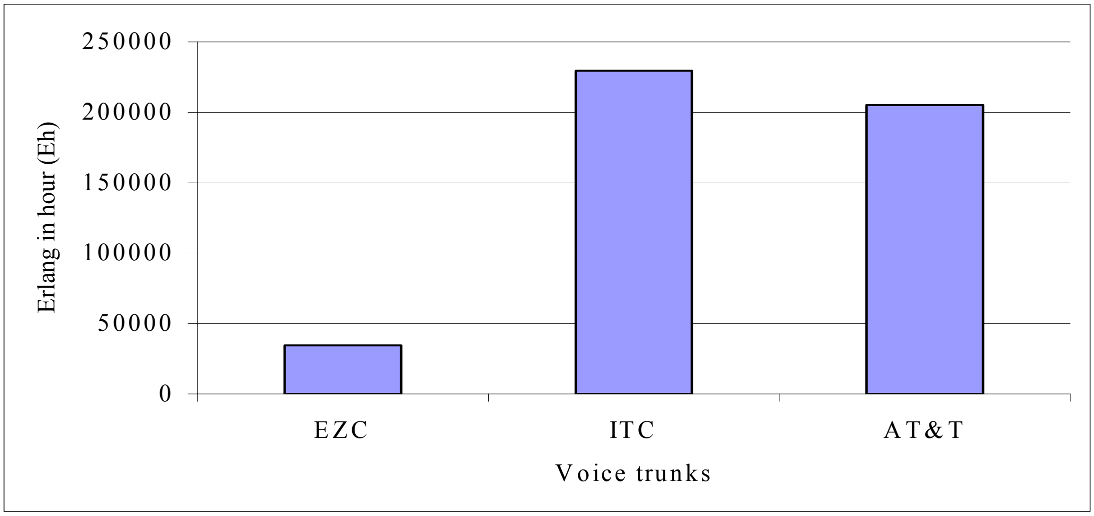
Graph (4.5.4): Mean ASR of EZC, ITC and AT&T Trunks

4.5.5 Carried traffic volume

The total volume of traffic carried during the 12 months, plotted in graph (4.5.5), is found as:

EZC trunk is	34421 Eh
ITC trunk is	229575 Eh
AT&T trunk is	205163 Eh

The traffic carried by VoIP trunks has reached 130% of that carried by the ordinary trunk.

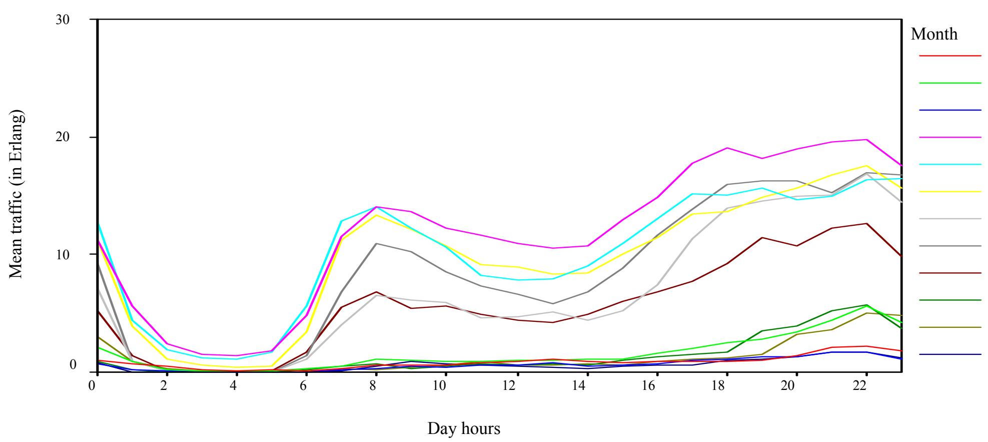


Graph (4.5.5): Carried Volume of Traffic in 12 Month

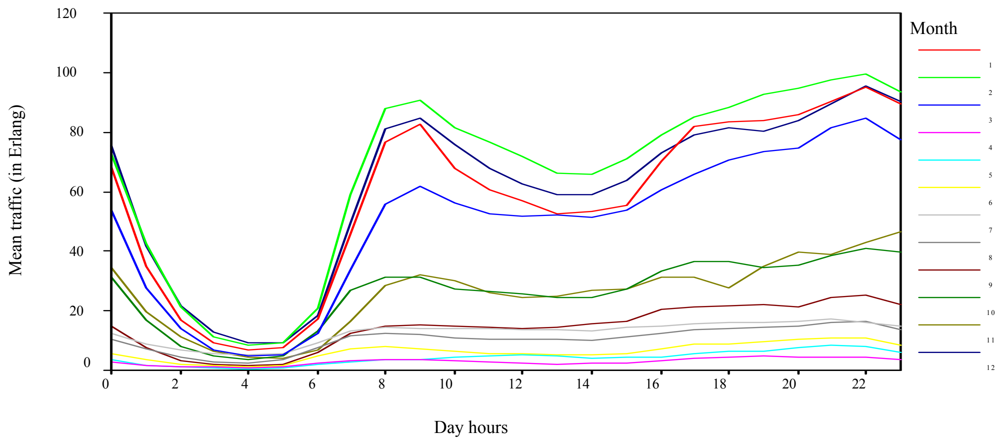
4.5.6 Traffic intensity behavior and grouping

The hourly mean traffic intensity behavior on monthly basis is plotted for EZC, ITC and AT&T in graphs (4.5.6.1), (4.5.6.2) and (4.5.6.3) respectively. The three graphs show drop in traffic during the early hours of the morning. The daily traffic profile is the same for all months.

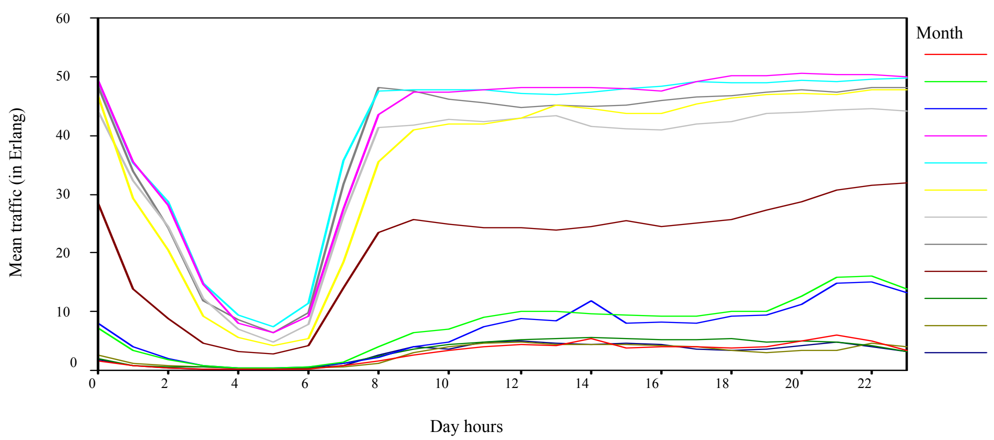
EZC trunk shows a recognizable degradation in traffic during the late months while ITC shows a considerable rise in traffic during the late months. AT&T shows a sever drop in traffic since last September 2002.



Graph (4.5.6.1): EZC Trunk Traffic Intensity Behavior on Monthly Basis

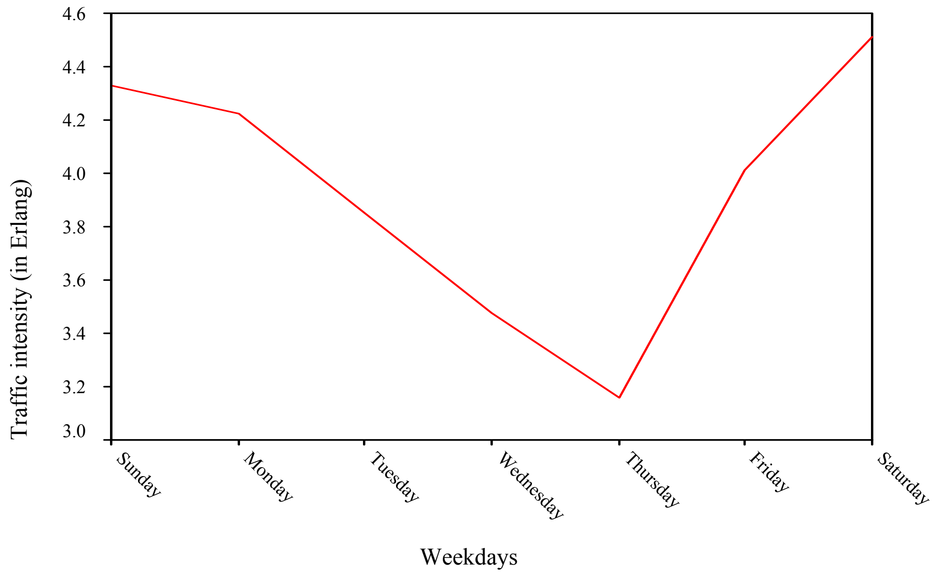


Graph (4.5.6.2): ITC Trunk Traffic Intensity Behavior on Monthly Basis

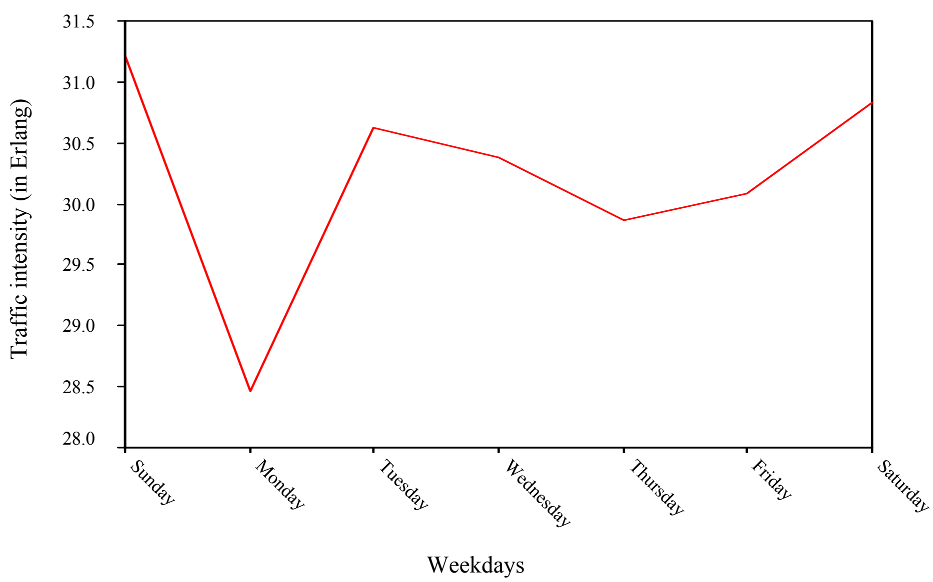


Graph (4.5.6.3): AT&T Trunk Traffic Intensity Behavior on Monthly Basis

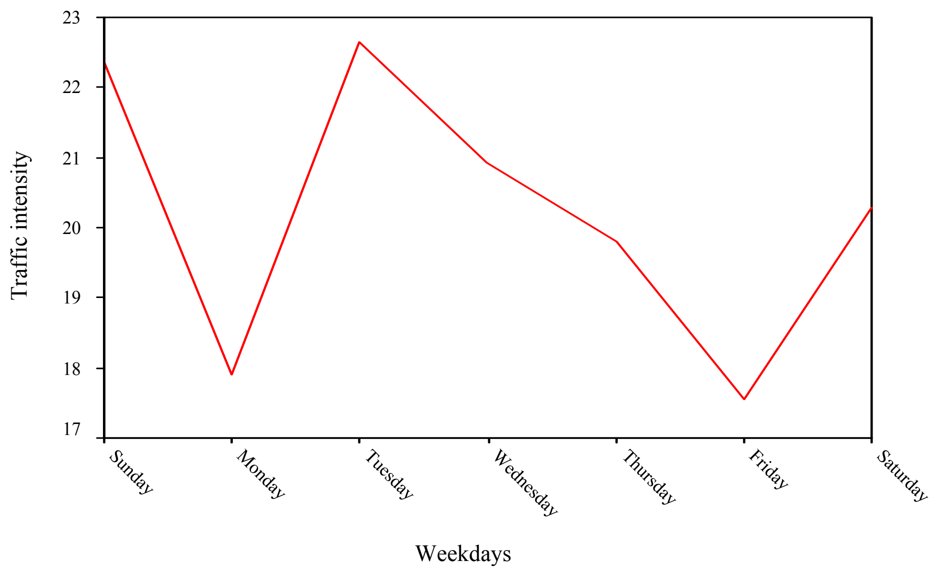
The traffic intensity behavior in the basis of weekdays are plotted for EZC, ITC and AT&T in graphs (4.5.6.4), (4.5.6.5) and (4.5.6.6) respectively.



Graph (4.5.6.4): EZC Trunk Traffic Intensity Behavior on Weekdays Basis



Graph (4.5.6.5): ITC Trunk Traffic Intensity Behavior on Weekdays Basis



Graph (4.5.6.6): AT&T Trunk Traffic Intensity Behavior on Weekdays Basis

5 CONCLUSIONS

The results of this study show that the implemented VoIP trunks has carried a total volume of traffic of 130% of that used to be carried by the ordinry trunks. The volume of the incoming-traffic carried by the ordinary trunks is dropped to 70% after introducing these VoIP trunks.

The implemented solution demonstrates that the establishment of VoIP trunk does not need any reconfiguration of the international satellite link. The system is directly connected to a port in the local Internet point of presence. The port provides a global reach to at ever ports in the Internet. There was no a big change in the platform and no complicated devices is installed.

The delay measurements shows that the delay in these VoIP trunks is within the recommended range.

The statistical analysis done on the carried traffic indicates that VoIP trunk has a daily traffic profile similar to that of the ordinary voice trunk. The high traffic load occurs at the same hours as for the ordinary trunk. This implies that the introduction of VoIP will not affect the current pricing policy of calls.

The traffic a pattern of VoIP has shown that there is always a room for other weekly or monthly varying patterns. Since VoIP trunks share the same international link, therefore VoIP trunks maximize the circuit utilization. Moreover, the statistics analysis has shown that the same analysis tools, procedures and results done to study ordinary voice trunks are applicable in VoIP trunks.

Calculation shows that VoIP trunks have better answer to seizure ratio and more average holding time than that of the ordinary trunk. Thus more paid time than that of the ordinay trunks could be expected.

The measurements of the IP-circuit utilization and the traffic volume determined that the drop in traffic in the ordinary trunks is more than one third, the growth in traffic generated by VoIP trunks is nearly 130%, and the utilization of the IP-circuits is less than 50%. These indicate that if one fourth of the bandwidth, which are currently used by the ordinary trunks, is changed into VoIP trunks then they are expected to be able to carry a traffic of towice the current incoming volume of traffic.

It can be concluded that the implementation of VoIP over the Internet in the international voice trunks will enhance the International link utilization and will extract a high volume of traffic with acceptable level of quality.

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