A Survey of QOS with IEEE 802.11e

Dr.P. Chenna Reddy[1]
Assoc. Prof. Dept. Of CSE
JNTUACE, Pulivendula
Y.S.R.(Dist).AP,INDIA

V. Venkata Ramana[2]
Asst.Prof. Dept of CSE
SSITS, Raychoty,
Y.S.R.(Dist).AP,INDIA

Dr A. Rama Mohan Reddy[3]
Professor, Dept. Of CSE
SVU College of Engineering,
Tirupati, A.P., INDIA

Abstract— IP is the fundamental protocol of Internet. It provides best efforts service. It has no in-built mechanisms to provide Quality of service. Some of the applications that are being used in Internet require Quality of service. IEEE 802.11 which is a wireless LAN standard doesn't support Quality of service. IEEE introduced the standard IEEE 802.11e to provide Quality of service. In this paper service differentiation mechanisms of 802.11e are studied. The limitations of it in providing Quality of service are identified. The scope for further research is presented.

Keywords- Application, quality of service, IEEE 802.11e, tuning.

I. INTRODUCTION

The Internet provides only best efforts service. Best effort service means, the service in which the network does not provide any guarantees in terms of the various performance parameters of the network like throughput, delay, packet loss, etc,. The network also does not differentiate sources of the traffic or the traffic itself (flow). In a best effort service, various elements of the network do their work to provide the best service possible. This may result in users obtaining unspecified variable bit rate and delivery time, depending on the current traffic load.

Applications for which the Internet is being used has changed over the years. The dominant applications of yesteryears, like File transfer, are not driving the Internet marker. Moreover the trend is towards online applications where the user interacts with the application without downloading it completely. Examples are audio and video streaming, game playing, etc. The significant characteristic of these applications is, they require guarantees in terms of network parameters like throughput, delay, packet loss etc, i.e., Quality of service (QOS).

QOS requirements of different applications [1] are as follows:

- **Web browsing**: The traffic of web browsing is bursty in nature. Bandwidth requirement is variable. Bandwidth requirement is low (when the user is looking at the web page) and suddenly bandwidth requirement becomes high (when the user selects a hyper link). It has low bandwidth requirement and typically value is 30 Kbps. It is sensitive to delay and the tolerable response time is below 5sec. It is not sensitive to jitter. Similar is the case with packet loss.
- **Email**: Response time for Email should be less than 5sec. No stringent requirements on delay and jitter. It has low bandwidth requirement and the typical bandwidth requirement is less than 10 Kbps. The expected loss rate and error rate are zero.
- **File transfer**: Response time for file transfer should be less than 5sec. It has high bandwidth requirement and the tolerance for loss rate and error rate is zero. It is not sensitive to delay and Jitter.
- **Interactive audio**: The typical bandwidth requirement is 64/128 Kbps. The delay should be less than 150msec and jitter should be less than 100msec. The response time should be less than 5sec. It is not sensitive to loss rate and error rate.
- **Interactive video**: The typical bandwidth requirement is 1.5Mbps. The delay and jitter requirements are same as Interactive audio. Response time is same as interactive audio and is not sensitive to loss rate and error rate.
- **Video conferencing**: The typical bandwidth requirement is 800 Kbps. The delay and jitter should be less than 150msec and 400msec respectively. Not sensitive to loss rate and error rate and response time for lip synchronization should be less than 100msec.

QOS is [2] measured from two perspectives: Technical perspective and User perspective. Originally, the notion of QOS was from the user perspective. ITU-T defines QOS as "the degree of satisfaction of a user of the service". In the course of time, the dominating research perspective on QOS has become more and more a technical one, focusing on monitoring and improving network performance parameters like packet loss rate, delay, jitter, etc,. Internet Engineering Task Force (IETF) defines QOS as a set of service requirements to be met by the network while transporting a flow. But end users usually are not bothered at all about technical performance; what they really care about is the experience they are able to obtain, and the Internet provided. From the end users perspective cost also plays a major role. User compares the services offered by different service providers. Moreover, different users have different requirements and hence different perspectives. Hence, there should be a mapping from user perspective to technical perspective. Recently, the term Quality of Experience has appeared, describing quality as perceived by the human user instead of as captured by technical network parameters.

To facilitate true end-to-end QOS on a wired network, the Internet Engineering Task Force has defined two models: Integrated Services (IntServ) and Differentiated Services (DiffServ). IntServ follows the signaled-QOS model, where the end-hosts signal their QOS needs to the network. DiffServ works on the provisioned-QOS model, where network elements are set up to service multiple classes of traffic with varying QOS requirements.

The IntServ model relies on the Resource Reservation Protocol (RSVP) to signal and reserve the desired QOS for each flow in the network. A flow is defined as an individual, unidirectional data stream between two applications. Two types of service can be requested via RSVP. The first type is a very strict guaranteed service that provides for firm bounds on end-to-end delay and assured bandwidth for traffic that conforms to the reserved specifications. The second type is a controlled load service that provides for a better than best effort and low delay service under light to moderate network loads. It is possible to provide the requisite QOS for every flow in the network, provided it is signaled using RSVP and the resources are available.

DiffServ, on the other hand, addresses the clear need for relatively simple and coarse methods of categorizing traffic into different classes, also called Class of Service, and applies QOS parameters to those classes. To accomplish this, packets are first divided into classes by marking the Type of Service byte in the IP header. A 6-bit bit-pattern (called the Differentiated Services Code Point) in the IPv4 Type of service octet or the IPv6 Traffic Class octet is used. When packets are classified at the edge of the network, specific forwarding treatments, formally called Per-Hop Behavior (PHB), are applied on each network element, providing the packet the appropriate delay-bound, jitter-bound, bandwidth, etc. This combination of packet marking and well-defined PHBs results in a scalable QOS solution for any given packet, and any application. In DiffServ, signaling for QOS is eliminated, and the number of states required to be kept at each network element is drastically reduced, resulting in a coarse-grained, scalable end-to-end QOS solution.

II. IEEE 802.11

IEEE 802.11 is the most widely used wireless LAN standard [3]. Developed as a simple and cost effective wireless technology for best effort services, IEEE 802.11 has gained popularity which make it de-facto wireless LAN standard. But it doesn't have built-in QOS support, and hence achieving QOS using it is a challenging task. It deals with two layers: Physical layer and Medium access control sublayer for wireless communications.

IEEE 802.11 MAC supports two basic medium access protocols: contention-based Distributed coordination function (DCF) and optional contention-free Point coordination function (PCF). PCF requires additional infrastructure in the form of Base point, which is technically called as Access point. Access point acts as a coordinator for medium access.

DCF is based on carrier sense multiple access with collision avoidance (CSMA/CA) instead of CSMA with collision detection (CSMA/CD), because wireless stations cannot listen to the channel for collision while transmitting. In IEEE 802.11, carrier sensing is performed at both physical and medium access control layers: physical carrier sensing and MAC layer virtual carrier sensing. DCF uses two parameters distributed inter frame space (DIFS) and contention window (CW) to control access to the channel. Before a station sends out a data frame, it senses the channel. If the channel is idle for at least a DIFS, the frame is transmitted. Otherwise, a backoff time slot is chosen randomly in the interval [0,CW). The contention window is incremented exponentially with an increasing number of attempts to retransmit the frame subject to a maximum of CWmax (contention window maximum). Upon receipt of a correct packet, the receiving stations waits a short interframe space (SIFS) interval and transmits a positive acknowledgment frame (ACK) back to the source station, indicating transmission success. During the backoff, the backoff timer is decremented in terms of slot time as long as the channel is determined to be idle. When the backoff timer reaches zero, the data frame is sent out. If collision occurs, a new backoff time slot is chosen and the backoff procedure starts again. After successful transmission, the CW is reset to CWmin (contention window minimum).

To reduce the effect of collisions, when long frames are used, request-to-send (RTS)/clear-to-send (CTS) mechanism is used. If the MAC frame length exceeds the RTS_threshold, RTS/CTS are used by stations. RTS/CTS mechanism allows virtual reservations by indicating to the remaining stations that a specific station wants to transmit for a certain time period. RTS/CTS contains a duration field which indicates the amount of time a station transmits. Two retry counters, the short retry count and long retry count, are defined for use in packet retransmission. Packets shorter than RTS_threshold are associated with the short retry count; others are associated with the long retry count. The retry counters begin at 0 and are incremented whenever a frame transmission fails. A frame is dropped if the retry count exceeds the maximum retry limit.

In order to optimize the performance [4] [5] [6] of DCF, parameters are tunable in both the physical and medium access control layers of 802.11. Some of the parameters are RTS_threshold, short retry count, long retry count, CW, and DIFS. However, these parameters are basically station-based and therefore cannot effectively differentiate multiple flows within a station. Furthermore, the effects of tuning these parameters are limited in terms of increasing/decreasing MAC throughput/ delay.

III. IEEE 802.11e

The IEEE 802.11e standard defines an additional coordination function called Hybrid Coordination Function (HCF) that is usable in QOS network. The HCF combines functions from the DCF and PCF with some enhanced, QOS-specific mechanisms and frame subtypes to allow a uniform set of frame exchange sequences to be used for QOS data transfers during both the CP and CFP. The HCF uses both a contention-based channel access method, called the enhanced distributed channel access (EDCA) mechanism for contention-based transfer and a controlled channel access, referred to as the HCF controlled channel access (HCCA) mechanism for contention-free transfer.

The EDCA mechanism provides differentiated, distributed access to the wireless medium for stations using eight different user priorities. The EDCA mechanism defines four access categories (AC) that provide support for the delivery of traffic with user priorities at the stations. Table 1, shows mapping from user priority to access category.

TABLE 1: User priority to Access category mapping

Priority	**User priority (same as 802.1D user priority)**	**802.1D designation**	**AC**	**Designation (informative)**
Lowest	1	BK	AC_BK	Background
·	2	--	AC_BK	Background
·	0	BE	AC_BE	Best Effort
·	3	EE	AC_BE	Best Effort
·	4	CL	AC_VI	Video
·	5	VI	AC_VI	Video
·	6	VO	AC_VO	Voice
Highest	7	NC	AC_VO	Voice

The differentiation in priority between ACs is achieved by setting different values for the AC parameters. IEEE 802.11e uses three parameters, which are as follows:

1) **Arbitrary inter-frame space (AIFS)**: It is the minimum time interval between the wireless medium becoming idle and the start of transmission of a frame i.e., the channel has to remain idle for AIFS time interval before the station transmits data.
2) **Contention Window (CW)**: A parameter used for random backoff. It is controlled by two configuration parameters CWmin and CWmax.
3) **TXOP Limit**: The maximum duration for which a station can transmit after accessing the channel without interruption.

The values of EDCA parameters are different for different Access categories. The values are set such the access category with higher priority gets better chance relative to access category with lower priority. AIFS value of high priority access category is low making it to access the channel quickly. The default AIFS, CWmin, CWmax, and TXOP values are as shown in Table 2. AC_BK, AC_BE, AC_VI, and AC_VO are access categories for background, best effort, video and voice traffic. The default values of CWmin and CWmax with physical layer using Frequency hopping spread spectrum are 15 and 1023 respectively. The default values of CWmin and CWmax with physical layer using Direct sequence spread spectrum are 31 and 1023 respectively.

TABLE 2: Default values of IEEE 802.11e parameters

AC	CWmin [AC]	CWmax [AC]	AIFS [AC]	TXOP [AC] limit (FHSS)	TXOP [AC] limit (DSSS)
AC_BK	CWmin	CWmax	7	0	0
AC_BE	CWmin	CWmax	3	0	0
AC_VI	(CWmin +1)/2-1	CWmin	2	6.016 ms	3.008 ms
AC_VO	(CWmin +1)/4-1	(CWmin +1)/2-1	2	3.264 ms	1.504 ms

The smaller AIFS value for a higher priority access category explains that the corresponding flow has to wait shorter time period before it can start transmission or counting down its bakeoff time compared to flow for a low priority access category. This ensures that higher priority access categories have quick access to the wireless channel. Quick access to the channel is particularly important for delay sensitive applications like VOIP (voice over IP). But at the same time, when the higher priority access categories have high traffic rate, the lower priority access categories will never get a chance which increases the delay and also decreases the throughput substantially. Also under high load conditions, high priority access categories suffer from high number of collisions which decreases the throughput of high priority access category. A smaller contention window for an access category will cause the corresponding flow to choose smaller random backoff values, and thereby waiting shorter time period in addition to AIFS as the medium becomes idle. Lower backoff values allow higher priority flows to quickly recover from collisions resulting in increase in throughput. Aggregate throughput and aggregate delay are better when service differentiation is achieved using CW differentiation keeping AIFS value same for all flows. Average End-to-End delay of higher priority flows is better with AIFS differentiation than CW differentiation.

But at the same time lower values for AIFS and contention window of higher priority traffic increases the probability of collisions. This is significant when higher priority flows traffic rate is high. As per the default settings of IEEE 802.11e parameters the CWmax of higher priority traffic is less than the CWmin of lower priority traffic. This allows higher priority traffic to quickly access the channel even after collisions. This may severely degrade the performance of the low priority access categories since they might not be able to decrement their backoff timers because of the smaller post backoff durations of the higher priority access categories. This problem is otherwise called as Starvation. In lightly loaded network conditions, the role of CWmax in introducing service differentiation is less significant compared to that of CWmin. It also indicates that in congested network situations, larger CWmax values for high priority ACs could remarkably reduce the collision rate.

TXOP parameter is not as significant as AIFS and CW parameters. It is suitable when the traffic is bursty. When the traffic is bursty, the corresponding higher priority access category can send the sequence of frames without any interruptions in-between. For constant data rate traffic it results in only little improvement in performance. The voice traffic is low data rate traffic, hence the performance of voice traffic doesn't improve much with TXOP. But it can improve the performance of video traffic since it requires high bandwidth. Differentiation based on only AIFS highlights the issue of coexistence of 802.11 and 802.11e, since the AIFS for high priority ACs in 802.11e is equal to DIFS in 802.11. However, as it is obvious, CWmin and CWmax are therefore used as additional differentiation parameters, i.e., smaller CWmin and CWmax values are used to introduce differentiation for high priority ACs in 802.11e.

IV. DYNAMIC TUNING OF IEEE 802.11e PARAMETERS

In [8] a new adaptive mechanism to adapt the CW to channel conditions is presented. Each station monitors the medium in order to measure the number of collisions for different data flows (video, voice, background and best effort data) and the number of successfully transmitted packets. With these values contention windows (one per AC) are split into different sub-windows and select one of these partitions on the fly, improving the performance under different load rates. The values of number of collisions and number of successfully transmitted packets are computed periodically. The limitation of this paper is, number of successfully transmitted packets and number of collisions are not perfect indication of the state of the channel or atleast not the only criteria for the status of the channel. Also in the process of computing new sub-contention windows, number of constants is used. Proper justification for the values of constants is not given.

The contention window of each priority level [9] is set after each successful transmission and after each collision. Setting the CW value is different after successful transmission and after collision. After successful transmission, the CW is set, assuming when a collision occurs a new one is likely to occur in the near future. The contention window is reduced slowly and is not reset to CWmin after successful transmission. This is done to avoid burst collisions. Instead of using static factor, the CW is reset adaptively using the estimated collision rate. Also to minimize the bias against transient collisions, an estimator of exponentially weighted moving average is used to smoothen the estimated values. The collision rate is calculated periodically. To ensure that the priority relationship between different classes is still fulfilled when a class updates its CW, each class uses different factor according to its priority level. After each unsuccessful transmission of packet of class i, the new CW of this class is increased with a Persistence Factor PF[i], which ensures that high priority traffic has a smaller value of PF[i] than low priority traffic. The assumption of another collision after a collision is valid with some probability. Though the probability is high, but it is not a certain event. Only collision rate is used as a factor for resetting CW after successful transmission. There are many other factors which are not considered.

The approach presented in [10] is based on adapting the values of CW depending on the channel congestion level. Based on the current value of CW, congestion level is estimated. The estimated link congestion ratio is calculated. To optimize further this ratio, a weight that reflects the certainty of the channel estimation is computed and the ratio is multiplied by the weight to get optimized value for ratio. The weight value is chosen as 0.9. Based on the value of ratio new value of CW is computed. The new value of CW would be a good representation of the backoff timer value needed for transmission for the current traffic priority taking into account the current network conditions. The metrics considered for evaluation are bit rate, end-to-end packet delay, and packet drop rate.

A new protocol, called Enhanced Distributed Coordination Function with Dual-Measurement (EDCF-DM), is proposed in [11]. EDCF-DM is based on the idea of reducing the number of idle slots by dynamically varying the contention window size according to the current traffic state of the traffic categories at each node. Meanwhile, it carefully adapts the contention window size based on the network condition of the system to avoid incurring extra collisions. When the workload of the system is high, CW is changed slowly to avoid further collisions. If other high priority traffic categories have low traffic, CW can be decreased faster to reduce the number of wasted idle slots.

The main performance impairment of distributed contention-based EDCA [12] scheme comes from packet collisions and wasted idle slots due to backoffs in each contention cycle. It tries to approximate the perfect scheduling among the queues. The perfect scheduling is one in which if a queue transmits, the probability that the others transmit should be equal to 0. It uses a backoff threshold value that separates two backoff states. Assuming that each priority queue acts as a virtual station, an adaptive backoff threshold is used for each priority queue. This allows to differentiate between the different priorities and to take into account the channel load which is necessary to increase further the total throughput of the medium. Basically, the backoff threshold should increase during low contention periods in order to reduce idle time, and it should decrease during high contention periods in order to reduce collisions. This is one of the main ideas behind this paper, which is adapting the backoff threshold for each class of traffic as a function of the channel load. To protect the quality of high priority flows without reducing the total throughput, adapted fast backoff mechanism is used which consists of increasing the size not only when there is a collision, but also when a queue senses the channel busy during deferring periods. This way, it is very likely that the highest priority flow will win the next channel access because of its smaller value compared to other flows.

In original EDCA, the problem is that a user only needs to wait for sufficient scattered idle time slots and then transmit after the back_off timer counts down to zero. Thus a low priority user may accumulate some idle time slots and may get the same privilege as a high priority user, which will result in higher collision rate. This is especially true if channel loading is very high, when an enormous amount of collisions cannot be avoided. The solution proposed in [13] is to adjust the contention window dynamically according to traffic load, which is based on the average collision rate.

Mechanism to support voice traffic in a mixed voice/data transmission over 802.11e WLAN is studied in [14]. The proper tuning of either AIFS or CWmin parameters can improve voice transmission quality at the wireless subnet while reducing the throughput of the background data traffic. It is also demonstrated that the quality differentiation with the AIFS parameter provides superior and more robust operation than access differentiation through the CWmin parameter. The AIFS differentiation is a superior mechanism to CWmin differentiation because of the existence of discrete instants of times where a lower number of stations may compete and access the channel. This increases the effectiveness of the overall random access mechanism for the high-priority stations. Tuning of the TXOP parameter does not improve the quality of voice transmission. This parameter plays an important role when large data comprising large packets sizes is to be sent. Since voice packets are short, TXOP parameter is of little use.

Several EDCA-based tuning algorithms [16] have been evaluated by comparing their flow-level response in presence of rigid (e.g. VoIP) and elastic (e.g. P2P) flows. Results show that those algorithms which adapt better to the changing WLAN state (number and type of active flows), and that are designed under multiple objectives, provide significantly higher performance and QOS than static and single objective configurations. Only two access categories have been considered, voice and best effort queues. The algorithms which use only one parameter and which use multiple parameters for tuning are considered.

V. Scope for further Research

1) Most of the experiments with 802.11e considered only CBR and UDP. Further research has to be done considering TCP as the Transport protocol. Also different types of traffic, particularly bursty traffic of WWW has to be considered.
2) The Adhoc network routing protocols are still evolving i.e., modifications are being made to the existing versions. Moreover, new Adhoc network routing protocols are being designed. Dynamic parameter tuning of IEEE 802.11e has to be experimented with different Adhoc network routing protocols.
3) Most of the previous experiments considered the primitive data rate 2Mbps of 802.11 physical layer. The data rate offered by 802.11 has changed drastically and higher data rates like 54Mbps are supported. Dynamic tuning of 802.11e parameters using higher data rates is required.
4) Various network parameters and traffic conditions to adjust the IEEE 802.11e parameters dynamically are to be considered.
5) Balancing high aggregate throughput and service differentiation of IEEE 802.11e has to be studied.
6) Voice requires low bandwidth. The IEEE 802.11e parameters have to be adjusted dynamically to provide each flow, only required QOS. QOS should not be at the risk of other flows or decrease in overall QOS.
7) There is need to investigate whether TXOP is really needed for voice transmission
8) IEEE 802.11 will coexist with IEEE 802.11e for some years to come. Hence there is need to study the impact of IEEE 802.11e parameters on IEEE 802.11.
9) Use of block acknowledgement scheme particularly when TXOP is used for data transmission has to be studied.

REFERENCES

[1] Yan Chen, Toni Farley and Nong Ye, “QoS Requirements of Network Applications on the Internet”, ACM Journal of Information-Knowledge-Systems Management, Vol. 4, Issue. 1, Jan. 2004.
[2] Emanuel Puschita, and Tudor Palade, “QoS Path Selection Mechanism in a Hybrid Access Wireless Scenario: A Distributed Approach using Mobile Agents”, International Journal of Communications, Vol. 2, Issue. 1, 2008.
[3] Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, IEEE Standard 802.11, 2007.
[4] Hua Zhu, Ming Li, Imrich chlamtac, and B. Prabhakharan, “A Survey of Quality of Service in IEEE 802.11 Networks”, IEEE Wireless Communications, Aug. 2004.
[5] Drabu, Yasir, “A Survey of QOS Techniques in 802.11”, Available: www.cs.kent.edu/~ydrabu/research/wmac/mac.pdf.
[6] Qiang Ni, Lamia Romdhani, and Thierry Turletti, “A Survey of QoS Enhancements for IEEE 802.11 Wireless LAN”, Wiley Journal of Wireless Communications and Mobile Computing, Vol. 4, Issue. 5, 2004.
[7] Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements, IEEE Standard 802.11e, 2005
[8] C. N. Ojeda-Guerra, and I. Alonso-Gonzalez, “Adaptive Tuning Mechanism for EDCA in IEEE 802.11e Wireless LANs”, 14th European Wireless Conference, 2008.
[9] Lamia Romdhani, Qiang Ni, and Thierry Turletti, “Adaptive EDCF: Enhanced Service Differentiation for IEEE 802.11 Wireless Ad Hoc Networks”, IEEE WCNC'03 (Wireless Communications and Networking Conference), New Orleans, Louisiana, USA, March 16-20, 2003.
[10] Joe Naoum-Sawaya, Bissan Ghaddar, Sami Khawam, Haidar Safa, Hassan Artail, and Zaher Dawy, “Adaptive Approach for QoS Support in IEEE 802.11e Wireless LAN”, IEEE, 2005.

[11] Hao Zhu, Guohong Cao, Aylin Yener and Allen D. Mathias, “EDCF-DM: A Novel Enhanced Distributed Coordination Function for Wireless Ad Hoc Networks”, ICC 2004 - IEEE International Conference on Communications, Vol. 27, No. 1, June 2004.
[12] Mohammad Malli, Qiang Ni, Thierry Turletti, and Chadi Barakat, “Adaptive Fair Channel Allocation for QoS Enhancement in IEEE 802.11 Wireless LANs”, IEEE ICC 2004 (International Conference on Communications), Paris, France, June 2004.
[13] Wen-Yen Lin, and Jung-Shyr Wu, “Modified EDCF to improve the performance of IEEE 802.11e WLAN”, Elsevier Computer Communications, Vol. 30, Issue. 4, 2007.
[14] Miroslaw Narbutt and Mark Davis, “The capability of the EDCA mechanism to support voice traffic in a mixed voice/data transmission over 802.11e WLANs - an experimental investigation”, 32nd IEEE Conference on Local Computer Networks, 2007.
[15] C. Cano, B. Bellalta, A. Sfairopoulou, and J. Barceló, “Tuning the EDCA parameters in WLANs with heterogeneous traffic: A flow-level analysis”, Elsevier Computer Networks, Vol. 54, 2010.

AUTHORS PROFILE

1. Dr. P Chenna Reddy is working as a Associate Professor of Computer Science and Engineering, JNTUACE.Pulivendula,AP INDIA. He Completed his Ph.D from JNTUH, Hyderabad. His areas of interest include Computer Networks and Mobile Computing. He has 14 years of Teaching experience.

2 V.Venkata Ramana obtained his Master’s degree in Computer Science Engineering and pursuing Ph.D degree in JNTUA, Anantapur and at present working as an Associate Professor in Department of Computer Science Engineering, SSITS, Rayachoti, Kadapa(Dist), A.P. His areas of interest include Software Architecture and Computer networking, data mining, and other latest trends in technology. He has more than 09 years of experience in teaching and research in the area of Computer Science and Engineering.

3. Dr.A Rama Mohan Reddy working as Professor of Computer Science and Engineering, Sri Venkateswara University.,Tirupati-Chittor-AP- INDIA. He Completed his M.Tech Computer Science from NIT, Warangal and completed his Ph.D in software Architecture from SV University,Tirupati. He has 30 years of Teaching experience.