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Junction Main Road
Suramangalam (PO)
Salem-636 005. TN. India
www.sonatech.ac.in

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Preface

The Communication Systems and Networks (CSN) is an inter-disciplinary group focusing on cutting-edge research in the development of reliable and efficient delivery of information for future Internet. It encompasses several areas of study including, but not limited to, telecommunication engineering, mobile communication, sensor networks, intelligent algorithms, network security and bio-inspired networks. The thrust of the research is in the development of intelligent protocols and architectures that offer seamless support for a variety of applications and user requirements in next generation networks. Work under this group includes algorithm design, protocol development and analysis, network programming, and prototype development. The main objective of the group is to establish a world-class collaborative research environment.

ANALYSIS OF VARIOUS VIDEO STREAMING ISSUES OVER HETEROGENEOUS WIRELESS NETWORKS

Ashwini – IV Year

Video streaming over wireless networks is an increasingly important and attractive service to the mobile users. Video streaming involves a large amount of data to be transmitted in real time, while wireless network conditions may vary from time to time. In this paper, a survey on different video streaming issues and analysis of proposed techniques for video streaming over Heterogeneous Wireless Networks is presented. This survey paper analyzes different schemes for video streaming i.e. rate allocation,bandwidth aggregation, multi user streaming and concurrent multipath transfer over heterogeneous wireless networks.

A vast majority of current wireless cellular networks are deployed using the homogeneous deployment scenario. The homogeneous cellular system is essentially a network base stations and user terminals with standards power level profiles and similar technical characteristics. All base stations in the network are similar and carefully planned for compatibility. This deployment scenario is complex, rigid, and expensive. Hence the need for a more flexible, cost-effective and ubiquitous deployment model capable of broadband delivery. This need informed the advent of heterogeneous networks, which allow for the deployment of non-homogeneous base stations, with the attendant advantage of improved spectral efficiency per unit area. Heterogeneous Networks are essentially made up of existing disparate Radio Access Network (RAN) technologies (e.g. WiMAX, Wi-Fi, E-UTRAN).

Video Streaming Issues

The evolution of wireless communication technologies requires the mobile clients (MC) to be equipped with multiple interfaces to access to different types of networks. To satisfy the user’s requirements with a better QoS the mobile client considers the characteristics of these access technologies, and decision making mechanism for data forwarding. This paper focused on the dynamic nature of the network conditions that satisfy multiple - often contradicting-constraints. Hence a fuzzy logic model has been proposed to translate the uncertainty factor to accurate values using the fuzzy logic tools. Then a throughput analysis of the metric values by various wireless technologies, and crisp values for the network parameters was derived. Along with throughput analysis, a sensitivity analysis based on the performance of each network was also carried out. The simulation results of this paper show that the best decision making mechanism for users to handle network uncertainty**.**

To maintain efficient utilization of the network resources by avoiding network congestion and to transport multiple video streaming sessions over a shared wireless network, careful rate allocation is needed. Hence a distributed scheme for congestion distortion with optimized rate allocation among various networks has been proposed in this paper. In rate allocation scheme for a given wireless channel the network congestion will increase with increase in allocated rate which leads to packet drop. On the other hand, decreasing the allocated rate leads to higher video distortion during encoding. At the application layer, the trade-off between average video quality and overall network congestion is optimized. As compared to TCP-Friendly Rate Control (TFRC), the proposed rate allocation scheme benefits in terms of both the video DR characteristics and wireless link capacities.

In heterogeneous wireless networks, Bandwidth aggregation is a challenging issue. For enhancing the video quality the throughput and reliability must be increased. To overcome the burst loss over heterogeneous wireless networks, a loss tolerant bandwidth aggregation approach (LTBA) has been proposed. The proposed LTBA is able to reduce the consecutive packet loss under burst loss assumption. To prove that the proposed LTBA outperforms the existing 'back-to-back' transmission schemes based on Gilbert loss model and continuous time Markov chain was carried out. Compared with the existing approaches LTBA shows improvement in terms of PSNR (Peak Signal-to-Noise Ratio) and improves the average video PSNR compared to the D-EMS, S-EMS and EDPF and guarantees 97 % of the video frames to be delivered within the decoding deadline. LTBA is able to handle the high loss channel with a loss probability of 40 %.

The problem of QoS (Quality of Service) provisioning for multi-user video streaming over multiple heterogeneous wireless networks based on the distributed, cross-layer design framework was considered. By jointly considering the rate allocation and the Joint Source-Channel Coding (JSCC), it aims at maximizing the QoS provisioning under the given resource constraint. At first a framework for optimal video rate allocation over multiple networks based on the observed Available Bit Rate (ABR) and the Round Trip Time (RTT) over each access network was developed. A distributed and cross-layer design was used to maximize the perceived video quality by combining rate allocation with joint source channel coding techniques.

The integration of multiple interfaces and a network layer architecture that enables diverse multi-access services was presented. An important aspect of the architecture when providing BAG services for real-time applications is the scheduling algorithm that partitions the traffic onto different interfaces such that the QoS requirements of the application are met. Earliest Delivery Path First (EDPF) algorithm was proposed, which ensures packets to meet their playback deadlines by scheduling packets based on the estimated delivery time of the packets.

Joint source-channel coding (JSCC) has proven to be an effective solution for video transmission over bandwidth-limited, error-prone wireless networks. However, one major problem with the existing JSCC approaches is that considering the network between the server and the client as a single transport link. To address the critical problem a novel flow rate allocation-based JSCC (FRA-JSCC) approach was proposed that includes three key phases: (1) forward error correction redundancy estimation under loss requirement, source rate adaption under delay constraint, and dynamic rate allocation to minimize end-to-end video distortion. Experimental results show that FRA-JSCC outperforms the competing models in improving the video peak signal-to-noise ratio as well as in reducing the end-to-end delay.

Under network bandwidth and playback delay constraints there is a problem of choosing the best streaming policy for multipath video delivery. The streaming policy consists of a joint selection of the network path and video packets to be transmitted, along with their sending time. A simple streaming model is introduced, which consists of the video packet importance, and the dependencies between packets. A careful timing analysis was needed to compute the quality perceived by the receiver for a constrained playback delay, as a function of the streaming policy. An optimization problem based on a video abstraction model, under the assumption that the server knows, or can predict accurately the state of the network was derived. Therefore a fast heuristic-based algorithm, based on load-balancing principles was proposed. Extensive simulations show that the proposed algorithm induces only a negligible distortion penalty. Simulation results also demonstrate that the proposed scheduling solution performs better than common scheduling algorithms, and therefore represents a very efficient low-complexity multipath streaming algorithm, for both stored and live video services.

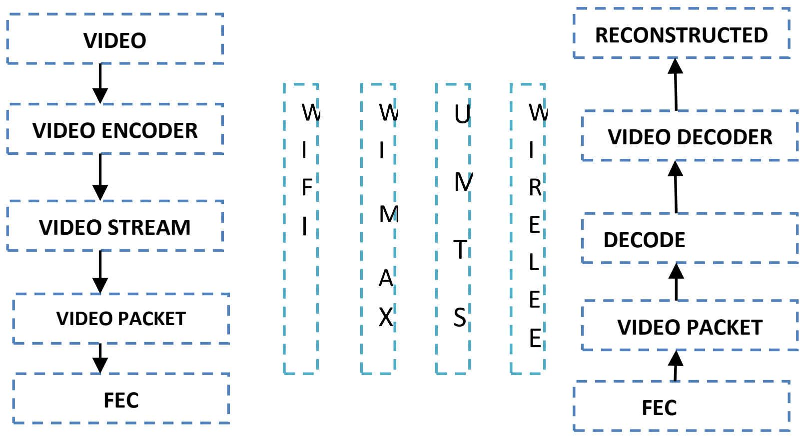
In wireless heterogeneous networks, data transmission and real-time video delivery plays a major role. To utilize the SCTP for FTP-like A novel Quality-aware Adaptive Concurrent Multipath Transfer solution (CMT-QA) was proposed. CMT-QA monitors and analyses regularly each path's data handling capability and makes data delivery adaptation decisions in order to select the qualified paths for concurrent data transfer. It includes a series of mechanisms to distribute data chunks over multiple paths intelligently and control the data traffic rate of each path independently. By reducing the reordering delay and unnecessary fast retransmissions CMT-QA's aims to mitigate the out-of-order data reception. In order to improve data delivery efficiency this policy differentiates between different kinds of packet loss and accelerates the retransmission if required. The simulation results demonstrate how the proposed CMTQA obtains better performance results for both reliable data transmission and real-time video delivery than classic SCTP CMT and CMT-PF mechanisms.

FEED FORWARD ALGORITHM FOR VIDEO STREAMING OF HETEROGENEOUS WIRELESS NETWORKS

Gowthami – IV Year

The Forward Error Correction technique and Super Forward Error Correction technique for correcting the errors and to overcome frame lost ant it is proven to be the best method to avoid frame lost in Heterogeneous Wireless Networks. Normally the high frame error rate decreases the quality of video streaming. The main advantage of FEC is that it does not require interaction with the video encoder and hence is applicable to any video coding technique, and to both stored and live video.

Block diagram



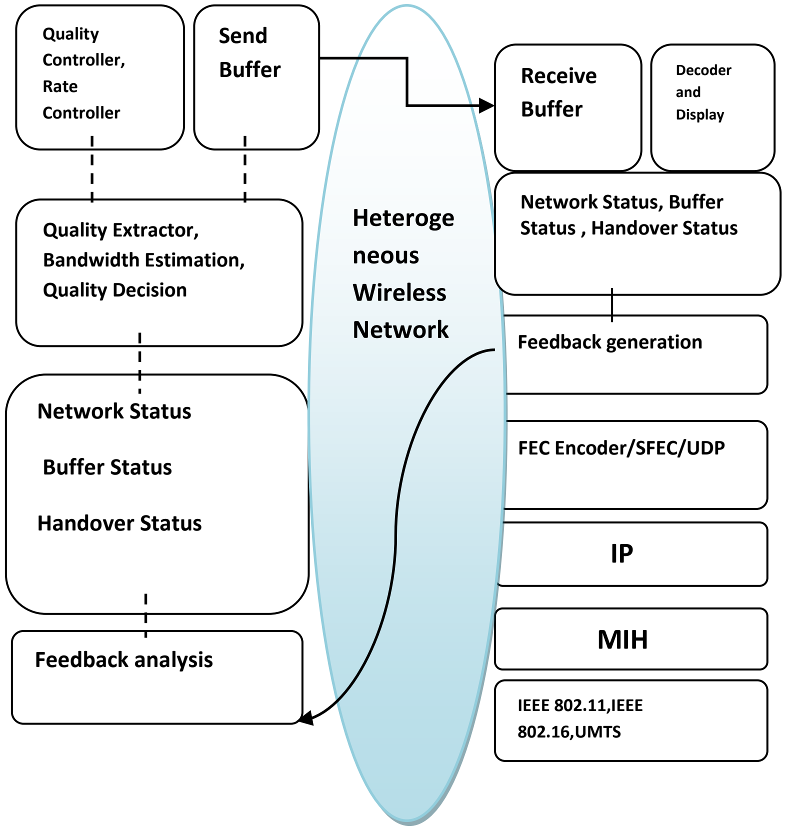
FEC for Video Streaming

FORWARD ERROR CORRECTION ALGORITHM STEPS

Step 1: Adding redundancy bits on compressed source bits to enable error detection and correction.

Step 2: Add a parity check bit at the end of a block of Video stream which can detect all single bit errors.

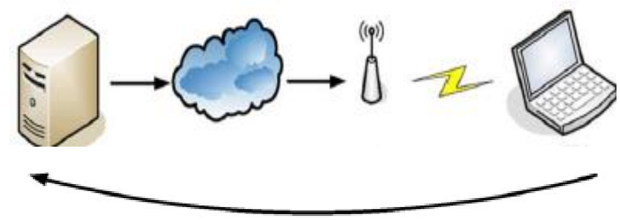
Step 3: For every k source bits add l channel bits to create n=k+l bits, channel coding rate r=k/n.



A Quality extractor module decides an optimal layer from the layered video bit stream based on the estimated bandwidth and the client's buffer status. Rate controller module appropriately adjusts its sending rate according to the varying network bandwidth. A Quality controller module also adjusts the quality level of the transmitted video stream based on the network bandwidth and the client's buffer status.

The streaming server and client together use sequence numbers for measuring the RTT. The header of the packet sent by the server includes a sequence number and a timestamp indicating the time when the packet is sent. The client sends the feedback packet to the server when the client received the server-side packet. Every time the client sends the feedback packet, it echoes the sequence number from the last received packet. A Bandwidth estimation module on the streaming server periodically estimates its available bandwidth using a TFRC throughput equation.

Video Sender Wireless N/w Video Receiver



Feedback Information [Frame count, Current buffer, Notification (HON), Flag]

Packet losses in wireless environments are generally recovered using either Automatic Repeat reQuest (ARQ) or Forward Error Correction (FEC) methods. ARQ schemes automatically retransmit the lost packets during timeouts, or in response to explicit receiver requests. Whereas in FEC schemes, the effects of potential packet losses are avoided by transmitting redundant packets together with the source packets such that a block of packets can be successfully reconstructed at the receiver end even if some of the packets are lost during transmission. While comparing the two approaches, FEC schemes result in a lower retransmission latency, and are therefore widely preferred for the delivery of video streams over wireless networks.

The basic principle of FEC mechanism is adding the redundant packets into the video stream together with the source packets such that packet losses can be recovered at the receiver end without the need for retransmission. Here the original block is encoded as packets, where it is the summation of source packets and redundant packets. Since FEC schemes enable the recovery of source packets which would be lost, where the effective loss rate in the transmission network is lower than the actual loss rate. In FEC coder, redundant packets are derived from the original packet using conventional coding theory techniques.

In this proposed scheme, the feedback information is periodically observed at the receiver side and any change in the network condition is fed back to the sender. Upon receiving this information, the sender calculates the round trip time (RTT) which further calculates the throughput using the formula. In other words, the FEC redundancy rate is dynamically controlled in such a way as to maintain a constant packet error rate at the receiver end.

RTT=feedback receive time-frame send time

Retransmission Timer Out (RTO)=4*RTT

Available Bandwidth=

$$S/ [(RTT * \sqrt{} ((2*B*P/3) + (RTT * 3) * \sqrt{}(3 *B *P)/8) *P *(1+32*P*P))]$$

Where

S-Size of Packet

P-Packet Loss Ratio (PLR) count

RTT-Round Trip Time

B-Feedback count

A REVIEW ON CHALLENGING ISSUES OF SELECTING NETWORK FROM HETEROGENEOUS WIRELESS NETWORKS

Sagaya Christina – IV Year

In heterogeneous wireless network environment, the major challenging issue is to select the network depends upon the demands from users and various networks. Network selection mechanisms play a major role to ensure the quality of service in the heterogeneous multi-access environment. In this research article, survey on different network selection mechanisms and analysis of various techniques for selecting a best networks are analysed for heterogeneous networks and given as key point for future researcher.

The construction of heterogeneous wireless networks consists of different types of wireless networks in user side and at the operator side; traffic from different access networks may share a common backbone bandwidth to provide various multimedia services. Hence bandwidth consumption of different network will reduce as shown in Fig. 1.

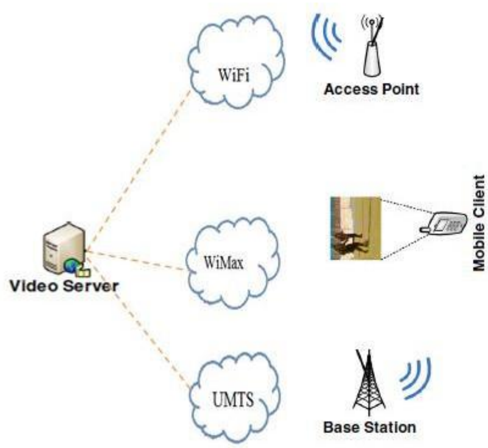


Fig. 1 Bandwidth Consumptions

Heterogeneous Networks are essentially made up of existing disparate Radio Access Network (RAN) technologies (e.g. WiMAX, Wi-Fi, UMTS, GPRS etc.). They consist of various architectures, transmission solutions, and base stations of varying power capacity. The constituent networks are used to improve the user experience, reducing bottlenecks in RAN and core network. Heterogeneous Wireless Networks are also used to introducing intelligent IP traffic routing and management, and also the efficient load balancing and resource allocation. Wireless LAN has been examined other inter-technology options. This is possibly to the attendant complementary offerings e.g. for WLAN: high data rates, low mobility and short range, while for UMTS: relatively low data rates, long range and high mobility. The service has been able to provide the ubiquity required in network coverage with accompanying Quality of Service (QoS) levels. This paper concentrates on various algorithms to select a best network in heterogeneous environment based on demands from users, cost and available bandwidth.

Network Selection Issues over Heterogeneous Wireless Networks

In communication research field it is very tough to select the best network for Heterogeneous Wireless Networks, and it is also a difficult to reduce the handoff and to provide service for all users at the same time. Hence a multiple attribute network selection algorithm to select the network based on Analytic Hierarchy Process (AHP) and synergetic theory was identified in various research areas. The algorithm takes both the coordinates of objective attributes and different QoS requirements into consideration. The network performance will be better if the synergetic value is greater. The entropy of the system is considered to be less if the synergetic degree is high. The synergic algorithm reduces the number of handoff and provides subscribers a better QoS according to different services.

Always Best Connected (ABC) scheme was recognized to guarantee the mobile users for network selection by integrating wireless local area network (WLAN) system, The ABC mechanism focused on the process of balancing user preference from different networks, service application based on the need of customer and network condition under traffic. ABC scheme comprises three parts: first the availability of WLAN was detected for providing best service, second an Analytic Hierarchy Process (AHP) was applied to calculate the relative weights of request from each users and third to normalize parameters and to calculate decision-making index to take decision for choosing best network. The advantage of the AHP technique is that it not only works for an integrated UMTS/WLAN system, but also be applicable to systems with more heterogeneity (e.g.WiMAX). Simulations reveal that AHP network selection technique can effectively decide the optimum network through making trade-offs among network condition.

A traditional way to select a target network based on the received signal strength (RSS) to meet the various demands of different multimedia applications and different users was introduced. The considered multiple criteria includes QoS, security, connection cost. Firstly, IEEE 802.21 was taken to obtain the information of neighbouring networks which falls into two categories compensatory information and non-compensatory information. Secondly, the non-compensatory information was used to sort out the capable networks. Thirdly, to combine a hybrid ANP and RTOPSIS model to rank the candidate networks the values of compensatory information was used as input values. Finally, a comparison study was made between TOPSIS based algorithm and new algorithm RTOPSIS based model to select best network.

The selection of the optimal access network requires service delivery in a heterogeneous wireless environment. The selection of an access network depends on several parameters such as the network and the application characteristics, the user preferences, and the service cost. An effective access network selection algorithm for heterogeneous wireless networks was designed that combines two Multi Attribute Decision Making (MADM) methods, the Analytic Hierarchy Process (AHP) method and the Total Order Preference. The AHP method is used to determine weights of the criteria and the TOPSIS method is used to obtain the final access network ranking. Hence the combination of these two methods can be very effective for the selection of the optimal access network according to requirements of the application.

In previous network selection policies were based on Shannon theory, they do not consider the delay characteristics. In reality, different services have different delay constriction. The two new network selection policies are introduced for heterogeneous environment systems using effective capacity which incorporate the delay constraint into the transmission rate. In this policies aim is to maximize the entire throughput with different delay constraints. The new network selection policies were updated which can efficiently improve network throughput and provide QoS guarantees for a delay constraints.

Mobile terminals in heterogeneous wireless environment the select the network within the initial access and handover process. Mobile terminal is connected to the network in possible way in terms of QoS performance and energy consumption. Before selecting a network the things which are to be taken into account are network condition, QoS performance and energy consumption. The best balancing network is to select based on the performance and energy consumption. Hence to improve energy consumption in real-time and non real-time applications, fuzzy logic was designed. The most difficult challenge faced by many researchers is to balance the process between different wireless networks for mobile devices is important to complete the handover process with successful network selection.

The handover operation is to find the delay and packet loss to the quality of service in a certain level. Selecting the best available network at the proper time is very important to the ubiquitous networks. Enhanced access router discovery (EARD) is illustrated to select the best network by prioritizing the network. EARD was developed and implemented to work in a heterogeneous wireless networks including of WiMAX and WLAN networks. The EARD method implemented to prioritize the rating for multiple criteria to select the target network and to evaluate the priority with respect to various conditions with different traffic types and one of the most challenging issues is to select the optimal network based upon the type of the demanding application. Vertical handoff will occur when a mobile terminal decides is to select the network from the available network. A network selection algorithm mainly based on Fuzzy Multiple Attribute Decision Making (MADM). The algorithm mainly consider these parameters Received Signal Strength (RSS), Monetary cost(C), Band Width (BW), Velocity (V) and user preference (P). MADM finds the Network selection function (NSF) that measures the efficiency in utilizing radio resources by handing off to a particular network. Network selection the highest NSF is considered to select the best network to hand off.

By analyzing the several network selection algorithms available at present for heterogeneous wireless networks. The network engineer can choose the suitable algorithm to provide good service for their customer and to achieve better results. This paper provides classifications of network selection issues involved in over Heterogeneous Wireless Networks and the techniques identified for researchers to solve their challenging issues. There are many unresolved issues addressed as future research.

Selection of Network in heterogeneous wireless Network using Fuzzy Logic

In this research, the network selection is based on the fuzzy logic in Multi Criteria Decision Making. A handover algorithm concentrates on making a decision based on incomplete information and in a region of uncertainty. Fuzzy logic is a multi valued function and allows intermediate values to be defined and it is suitable for uncertainty.

The method employed here is the fuzzy logic in multi Criteria decision making. Compared to other methods, the mathematical concepts are quite simple and easy in fuzzy reasoning. The information, data and the measurements of the input parameters are dissimilar and obtained from different sources in other methods where as in fuzzy this problem won't exists.

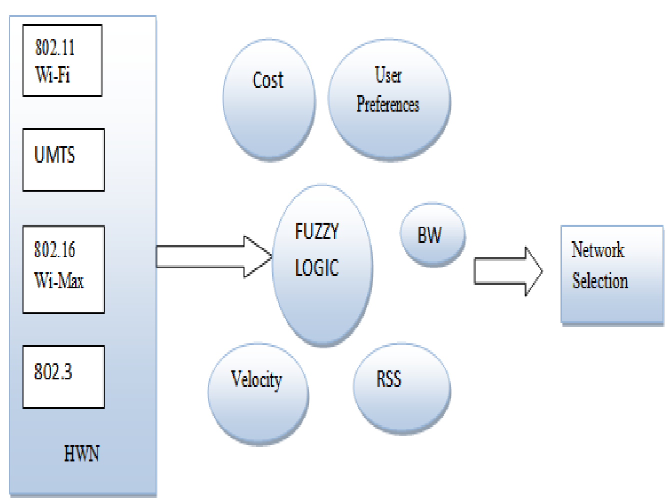


Fig:2 Network Selection using Fuzzy Logic

Multi-objective function is developed to select the network in a heterogeneous wireless environment. The multi-objective function consists of dominant parameters such as delay, bandwidth, packet loss and cost. These parameters particularly characterise the performance of voice, video and data services.

The multi objective function is the summation of the product of the normalised weight of the parameter and average value of the parameters. In general the normalised weight of the parameter is equal to unity. With the help of this function the network can be selected.

TOPSIS method is a multi criteria decision analysis method which is based on the concept that value should be taken which have a shortest distance from the positive ideal solution and longest distance from the negative ideal solution. This method considers some parameters, identifying its weights, normalising those weights and finally calculating the distance between each value and the ideal value.

The network selection process is carried out using the heterogeneous wireless network which cumulatively serves as base station. Heterogeneous network is the use of multiple types of access nodes in a wireless network. Here we are considering four wireless networks: UMTS, 802_3, 802_11, 802_16. From this base station network it gets implemented by performing fuzzy logic technique.

Fuzzy Multi Criteria Decision Making method is used to select the best network in heterogeneous environment. Selection of the best, from a set of alternatives, each of which is evaluated against multiple criteria. Fuzzy logic is a multi valued logic and allows intermediate values to be define. Provides an inference mechanism which can interpret and execute command. A handoff algorithm must be capable of making a decision based on incomplete information and in a region of uncertainty.

We are designing an adaptive multi criteria handoff decision algorithm that incorporates fuzzy logic because of the inherent strength of fuzzy logic in solving problems exhibiting imprecision and the fact that many of the terms used for describing radio signals are fuzzy in nature.

While implementation of this technique certain parameters should be considered they are cost, bandwidth, received signal strength, velocity and user preference. Therefore finally based on this implementation and performance analysis the network selection process is carried out.

IMPLEMENTATION FUZZY LOGIC TECHNIQUE

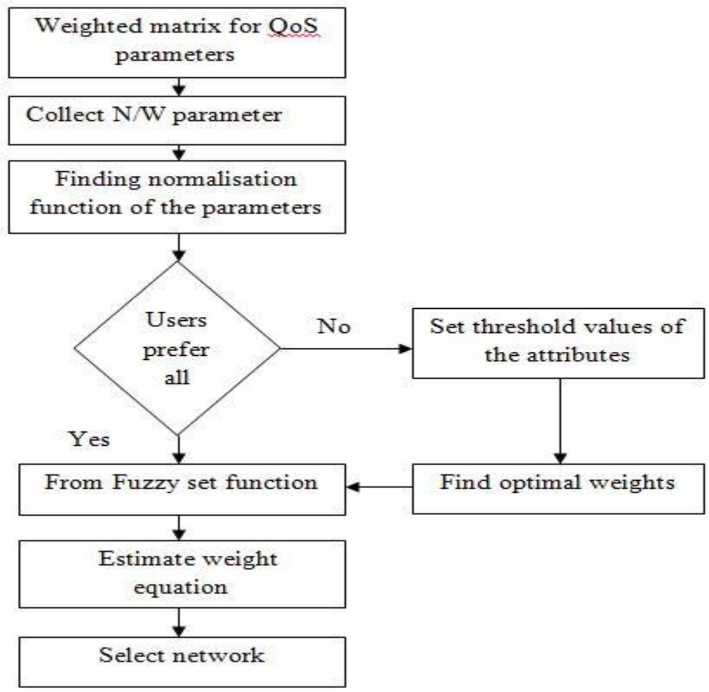


Fig:3 Flow chart for network selection using MCDM

The networking leads to the issue of handover, thus an effective and efficient handover process is important to select the user's connection from one cell to another cell, when the user moves from one location to another. In heterogeneous wireless environment the following four wireless networks are considered: UMTS, 802_3, 802_11, 802_16.

The multi-objective function is defined for the QoS parameters. The normalised weights are assigned for all the parameters. The sum of normalised weight must be equal to unity. Weight coefficients are calculated by finding the entropy of attribute. Then the network parameters such as bandwidth, cost, and received signal strength, velocity and user preference are collected. The normalized values of the above parameters are found.The fuzzy set function is collected. If the users prefer all the parameters equally form a fuzzy set or else set some threshold to optimize the parameters. Atlast by using the fuzzy set, select the suitable network.

Vertical Handoff Decision Algorithm for Heterogeneous Wireless Networks

One of the major design issues in heterogeneous wireless networks is the support of vertical handoff. Vertical handoff occurs when a mobile terminal switches from one network to another (e.g., from WLAN to CDMA 1xRTT). The objective of this paper is to determine the conditions under which vertical handoff should be performed. The problem is formulated as a Markov decision process. A link reward function and a signaling cost function are introduced to capture the tradeoff between the network resources utilized by the connection and the signaling and processing load incurred on the network.

A stationary deterministic policy is obtained when the connection termination time is geometrically distributed. Numerical results show good performance of our proposed scheme over two other vertical handoff decision algorithms, namely: SAW (Simple Additive Weighting) and GRA (Grey Relational Analysis).

SURVEY ON QUALITY OF SERVICE PROVISION IN 4G WIRELESS NETWORKS

Rebekha – IV Year

Worldwide many mobile operators, industry experts, and researchers have diverse visions of potential 4th generation (4G) features and its implementations. 4G networks will be incorporating advanced Internet Protocol version 6 (IPv6) protocol and the signaling will be done through Internet Protocol (IP). There are several key challenges in implementing 4G heterogeneous network. Few of these problems are all IP network, integration across different topologies, security and Quality of Service (QoS). This paper gives a survey and classification of the important QoS approaches proposed for 4G networks. Classification is based on the work done in each protocol layer and Cross Layer Design (CLD) approach. Finally, this paper presents outcomes of survey which includes significant observations, limitations and idea of further research in improving QoS in 4G networks.

The requirement for higher data speed is increasing rapidly, reason being the availability of smart phones, at low cost in the market due to competition and usage of social networking websites. Constant improvement in wireless data rate is already happening. Different network technologies are integrated to provide seamless connectivity and are termed as heterogeneous network.

Long Term Evolution-Advanced (LTE-A) is known as 4G and it is the solution for heterogeneous networks and wireless broadband services. International Mobile Telecommunication-Advanced (IMT-Advanced) represents a family of mobile wireless technologies, known as 4G.

Network evolution is occurring throughout the globe and we are shifting toward an all-IP communications. The core of 4G network is IP and the signaling is done through advanced IPv6 itself. Internet Protocol (IP) describes the format as well as the switching technology of what is popularly called Evolved Packet Core (EPC). Basically IP was termed as a general-purpose data transport protocol in the network layer, but now extended as a carrier for voice and video communications over 4G networks.

Wireless networks in the future will be heterogeneous. Different access networks such as Institute of Electrical and Electronics Engineers (IEEE) 802.15 Wireless Personal Area Network (WPAN), IEEE 802.11 Wireless Local Area Network (WLAN), IEEE 802.16 Wireless Metropolitan Area Network (WMAN), General Packet Radio Service (GPRS), Enhanced Data rate for GSM Evolution (EDGE), Wideband Code Division Multiple Access (WCDMA), Code Division Multiple Access (CDMA2000), satellite network etc are integrated. Selecting the suitable access network to meet the QoS requirements of a specific application has become a significant topic and priority is to maximize the QoS experienced by the user. QoS is the ability of a network to provide premier service to some fraction of total network traffic over specific underlying technologies. QoS metrics are delay, jitter (delay variation), service availability, bandwidth, throughput, packet loss rate. Metrics are used to indicate performance of particular scheme employed. QoS can be achieved by resource reservation (integratedservices), prioritization (differentiated services). We can apply QoS according to per flow (individual, uni- directional streams) or per aggregate (two or more flows having something in common) basis. From the QoS point of view, the protocol stack is composed of upper layer protocols (transport and above), on top of IP. Applications can, in this context, be classified according to the data flows they exchange as elastic or real-time. The network layer includes IP traffic control that implements datagram policing and classification, flow shaping, and scheduling. The data link layer may also provide QoS support, by means of transmission priorities or virtual channels. QoS provision in 4G networks is challenging as they support varying bit rates from multiple users and variety of applications, hostile channel characteristics, bandwidth allocation, fault-tolerance levels, and frequent handoff among heterogeneous wireless networks.

QoS support can occur at the network, transport, application, user and switching levels. To meet QoS, we should address the following issues like encryption protocols, security and "trust of information", different rates, error profiles, latencies, burstiness, dynamic optimization of scarce resources and fast handoff control. Over the past several years there have been a considerable amount of research in the field of quality-of-service support for 4G systems as it‟s more challenging than previous generations. Regarding this, some research papers have presented their idea of QoS architectures across all protocol layers. QoS solutions proposed for 4G network can be classified based on the layer in which the mechanism works. Although research to provide QoS in 4G network has happened in data-link, physical, transport and application layer, predominant architectures are available in network layer. A different approach is cross layer design for providing QoS in 4G networks where it tries to optimize architecture across adjacent layers. Traditional approach has been to treat the layers as different entities. A higher layer protocol only makes use of services at lower layers and is not concerned about the implementation of service.

In CLD approach, protocols can be designed by allowing direct communication between entities in nonadjacent layers for resource optimization. CLD in wireless is different mechanism than CLD in wireline. An IP-Based QoS Architecture which supports multiple access networks and multiple service provider scenarios. It is an integrated management approach to service in the case of heterogeneous network. Mobile network access is based on the association between QoS brokers and Authentication, Authorization, Accounting and charging systems (AAAC). QoS signaling architecture which integrates resource management and mobility management is also presented. Architecture is developed with the concept of domain resource manager and capable of supporting various handover types. Few approaches consider core issues in the design of QoS mechanism. But they fail to provide a fully integrated QoS approach to IP-based communication for variety of applications and protocols. Usually adaptive applications are disregarded and mobility issues are not taken care.

THROUGHPUT PREDICTION-BASED APPROACH

Pavithra – IV Year

Throughput can be considered one of the most intuitional parameter for indicating the states of networks. Thus, a number of early studies developed the bit rate adaptation schemes based on measuring throughput or its prediction. For example, argued that the key for adaptive streaming was identifying the network congestion and measuring the network throughput.

Nonetheless, in a practical system, it is difficult to differentiate whether the throughput variation is from the short-term TCP congestion control or the physical networking environment. Thus, a bit rate adaptation scheme was presented based on bandwidth changes, which is estimated using a smoothed HTTP throughput based on the segment fetch time and analyzed many commercial video players and found that existing adaptive streaming schemes consist of three key components: throughput prediction, bit rate selection, and video chunk scheduling. Thus, a series of algorithms (called FESTIVE) were proposed to optimize the decisions in these three components with the goal of fairness, stability, and efficiency.

Buffer Occupation-Based Approach

In addition to throughput, buffer occupation is another efficient metric for adaptive streaming in existing studies. The basic idea is that the adjustment should make the bit rate selection more conservative when the buffer is at risk of under running and more aggressive when the buffer is close to full. For example, argued that it could be challenging to pick the appropriate video bit rate by estimating future network capacity in an environment with highly variable throughput.

In response, an alternative approach BBA (buffer based algorithms) was proposed to directly choose the video bit rate based on the current buffer occupancy. The investigation reveals that throughput estimation is unnecessary in steady state; however using simple estimation (based on immediate past throughput) is important during the startup phase, when the playback buffer is growing from empty.

In addition, formulated the bit rate adaptation as a utility maximization problem through the static control algorithm based only on the buffer occupation. In particular, an online control algorithm called BOLA (Buffer Occupancy based Lyapunov Algorithm) was proposed to minimize re buffering and maximize video quality using stochastic optimization techniques.

Integrated Information-Based Approach

In addition to the prior two approaches, several recent works achieved bit rate selection based on the integrated information of throughput prediction and buffer occupation simultaneously. Generally speaking, these approaches select the appropriate bit rate mainly based on the throughput, and then make appropriate adjustments according to the buffer occupation information. For example, demonstrated the limitations of the traditional throughput prediction-based approach, and presented a client-side rate adaptation algorithm PANDA (Probe AND Adapt).

The fundamental operation of PANDA is similar to TCP congestion control protocol. The difference is that the congestion is detected via the throughput degradation, not the traditional packet loss ratio or end-to-end delay and argued that the client-side adaptive algorithm could generate an on-off traffic pattern, which will lead to unfairness and low bandwidth utilization when many video flows share a bottleneck.

In response, a client-side controller ELASTIC (feedback Linearization Adaptive Streaming Controller) was presented to address this problem using feedback control theory. The formulated the video bit rate adaptation as a stochastic optimal control problem and developed a model predictive control (MPC) algorithm, which combines both throughput prediction and buffer feedback signals. The formulated the streaming adaptation as an optimization problem to maximize the long-term QoE by considering video quality, quality switching frequency, and starvation events. The optimization problem was solved by breaking it into resolvable sub-optimization problems of each segment to meet the real time constrains. Although many efficient algorithms have been presented for the adaptive streaming problem, there is still significant space for QoE improvement. First, the performance of information prediction-based approaches relies heavily on the prediction algorithm. Nonetheless, the accurate prediction is quite difficult in a network with dramatic throughput fluctuation, such as mobile wireless networks with heterogeneous deployment. Second, the buffer occupation-based approaches depend on the size of the buffer pool.

Small buffer size many lead to problems of frequent bit rate jitter and TCP congestion in on-off traffic patterns. In addition, because QoE is a purely subjective measure of the overall acceptability from the user’s perspective, the principle of defining QoE and the corresponding bit rate adaptation should take the overall tradeoff and personalized customization into consideration. Our study differs from the existing work in the following aspects: First, our study focuses on mobile video streaming in next-generation wireless access networks with heterogeneous network deployments, in which various wireless infrastructures provide dynamic network accesses. Specifically, the network throughput is rapidly varying, and is difficult to predict due to the mobility of user terminals. Second, the dynamic control scheme proposed in this paper is more practical and can be performed by the client-side device based on the measurable system states without any prediction of future throughput.

CONGESTION CONTROL IN THE HETEROGENEOUS WIRELESS NETWORKS

K.Manikandan – IV Year

As the Internet evolves, TCP also faces many new challenges posed by emerging Network technologies. A large proportion of these challenges arise from the ever-growing Popularity of wireless networks, driven by the users demands for un tethered Internet access. Compared to their wired counterparts, wireless networks have the following major advantages:

1. The deployment of wireless networks is very easy because tedious configuration, civil engineering infrastructure and placement planning procedures, which are usually needed in wired networks, can be to a large extent avoided.
2. Under certain circumstances wireless Internet access is more economical than the wired approach. For example, in remote villages with no wired network infrastructure, it is difficult to access the Internet. In this case employing a satellite link would be cheaper and more effective than deploying wires from the only available ISP center far away.
3. Wireless networks can provide unconstrained network access, with possible user mobility. For example, in cellular networks users can stay connected even on a fast moving vehicle.

In a nutshell, the emergence of wireless networks, including satellite Internet links, wireless hot-spot (WLAN), mobile ad-hoc network, 3G cellular network, high speed down-link packet access (HSDPA), and so on, has made the Internet a wiredcum- wireless communication platform: while the backbone of the network relies on wired, often optical links, access networks are relying more and more on wireless technologies.

Despite the advantages they bring about to Internet communication, wireless links suffer also from many disadvantages compared to their wired counterparts. Wireless communications take place in an open transmission medium which is subject to various environmental interferences and noises. In addition, the same wireless spectrum is often shared by many wireless nodes, resulting in multi-user interference and collisions.

Due to various factors, such as, the interference, channel fading, hidden nodes, user mobility, and so on, non-congestion-related packet losses are very frequent in wireless networks. Furthermore, while wired network bandwidth evolved homogeneously across all the domains of the network3, wireless networks bandwidth varies across a large spectrum and can be as narrow as several kbps in GSM cellular network, or as broad as hundreds of Mbps in wireless metropolitan area network (WMAN). This contributes significantly to increasing the bandwidth-delay product in the network.

Finally, due to handoff and other factors, there could also be temporary disconnections in wireless networks, which affect the operation of the TCP congestion control mechanism. To effectively utilize network resources and achieve good fairness in such a heterogeneous wired-cum-wireless environment, new challenges are posed to the conventional TCP congestion control mechanisms.

In this research various TCP congestion control issues in the wiredcum- wireless Internet, review some of the challenges, and survey the related literature. We do not intend to present a comprehensive list of all the proposed schemes; instead, we provide a summary of major research efforts and highlight the representative approaches to address different issues. We categorize the state-of-the-art research progress with respect to the major issues faced by TCP congestion control in the wired-cum-wireless Internet.

CHALLENGES AND PROSPECTS OF WIRELESS TECHNOLOGY

Meenakshi – IV Year

With the IMT-Advanced (IMT-A) standards ratified by the International Telecommunications Union in November 2010 and IMT-A, i.e., the fourth generation (4G), wireless communication systems being deployed in the world, the fifth generation (5G) mobile and wireless communication technologies are emerging into research fields.

Based on the Internet Protocol Architecture of 4G communication systems, unprecedented numbers of smart and heterogeneous wireless devices will be accessing future 5G mobile and wireless communication systems with a continuing growth of Internet traffic. Therefore, compared to 4G communication systems, significantly higher wireless transmission rates are expected in 5G communication systems, such as 10 Gbps peak data rates with 8~10 bps/Hz/cell.

Moreover, energy efficient concepts will be fully integrated into future wireless communication systems to protect the environment. To meet the above challenges, 5G mobile and wireless communication systems will require a mix of new system concepts to boost spectral efficiency, energy efficiency and the network design, such as massive MIMO technologies, green communications, cooperative communications and heterogeneous wireless networks. We expect to explore the prospects and challenges of 5G mobile and wireless communication systems combining all of the above new designs and technologies. Thus concluding, simultaneous management of multiple technologies in the same band limited spectrum is a challenge in 5G mobile communication which supports going beyond voice for newer smart phones and advanced mobile devices. Gathered data for meeting the requirements and satisfactory constraints are highly valuable for the development of 5G cellular communications at mm bands in the coming decade.

Challenges of OFDM

There are severe practical deployment and performance issues with OFDM (notwithstanding all the positives mentioned above), which limit its usability in the use-cases which are deemed important for 5th generation cellular networks. Over the last few years, OFDM deployments have been slowly gathering steam, both in WiFi and in LTE networks. Gradually, a corpus of data is building up real-life performance in the field, which has provided designers with further insight into the strengths and drawbacks of the technology.

OFDM is based on orthogonality of sub-carriers; this requires strict control of the synthesizer. From the perspective of a single transmitter, this is easy to achieve. However, as discussed above, 5G networks will require tight inter-coordination between multiple network elements and tight sharing of spectrum.

In certain applications where a subset of subcarriers is allocated to each user, such as cognitive radios and multiuser multicarrier systems, OFDM isn't quite an optimal solution. OFDMA can only operate if strict time and frequency synchronization between users and a base station is achieved. Distributed transmission techniques, such as CoMP, impose an even more stringent requirement of synchronous base stations on top of everything else.

As of now, synchronizing cooperative networks is expensive, and in some cases even impossible. The problem only becomes tremendously more challenging when considering low-cost micro and femto base station deployments or even lower-cost sensor devices, which may transmit without synchronizing to the network's clock.

Another critical issue in CoMP-OFDM systems is their sensitivity to multiple carrier frequency offsets (CFOs) between terminals and base stations. The frequency offset can be caused by either doppler shift as a result of terminals' mobility or by oscillator frequency mismatch between a transmitter and a receiver. Multiple CFOs in CoMP-OFDM systems destroy the orthogonality between OFDM subcarriers and causes intercarrier interference (ICI) at the receiver, which leads to significant degradation in system performance.

There is also a need to address the spectral efficiency aspect, as future 5G networks will have to use scarce and fragmented spectral resources. It poses extreme challenges when it comes to achieving blocking requirements and satisfying regulatory out-of-band spectrum constraints which cannot be achieved by the spectrum shape of OFDM (OFDM waveforms inherently have large side-lobes because of the rectangular shaping of the temporal signal).

There is a need for a filtered, multicarrier approach with reduced side-lobe levels of the waveform which could minimize inter-carrier interference (ICI).

Filter Bank Multi-Carrier

FBMC (filter bank multi-carrier) generalizes traditional orthogonal frequency-division multiplexing (OFDM) schemes, allowing a nonrectangular sub-channel pulse shape in the time domain. This approach leads to a better spectral containment that improves interference mitigation in several time-variant environments. The key idea behind this technique is to perform a nonrectangular pulse-shaping as compliant as possible with the channel characteristics (time and frequency dispersion). From a transmission perspective, the FBMC technique has the potential to increase bit rate, due to the reduced guard bands and the absence of the cyclic prefix needed in OFDM. FBMC also allows the possibility to allocate different subcarriers to different unsynchronized users in a spectrally efficient manner. The out-of-band emission of FBMC is much lower than OFDM.FBMC only becomes efficient with offset

QAM (OQAM), where the real part of QAM symbols are mapped to one half of the multi-carrier symbols and the imaginary parts are mapped to an interlaced half of the multi-carrier symbols. While this works well with single-cell, single user transmission, in the JR case, we obtain additional interference paths between the interlaced OQAM symbols. Furthermore, certain types of MIMO transmission are not supported by FBMC/OQAM.

Universal Filtered Multi-Carrier

A paramount feature required in future wireless communication systems, supporting the Internet of Things (IoT) and Massive Machine Communication (MMC), is to efficiently support transmission of small data packets. A physical layer enabling this target demands efficient support of short transmission bursts. Here, FBMC/OQAM with its long filter lengths by design loses efficiency. Thus an alternative modulation scheme to FBMC is needed, where a filtering operation is applied to a group of consecutive subcarriers instead of the per subcarrier filtering used in FBMC. By using the technique called universal-filtered multi-carrier (UFMC), the effect of side-lobe interference on the immediate adjacent sub-channels is significantly reduced. This offers better ICI robustness and better suitability for fragmented spectrum operation. The UFMC technique uses shorter filter lengths compared to OFDM’s cyclic prefix lengths, making it applicable for short burst communication. The UFMC technique can be considered as a potential candidate for future wireless systems which have to support a plethora of low-cost devices (IoT and MMC).

IWLAN FOR INDUSTRIAL AND INDUSTRY-RELATED AREAS

Neha Prakash Kacholiya – IV Year

IWLAN is especially suitable for demanding industial applications that require end-to-end, reliable and secure radio communication:

- For implementation at industrial and automation customer sites
- For outdoor environments with demanding climatic requirements
- For low-cost integration in the control cabinet or in devices.

Thanks to the use of mobile devices linked via wireless data networks (e.g. wireless LAN), the efficiency of processes can be significantly improved. The primary benefit of wireless solutions is the simple and flexible availability of mobile or hard to access stations.

Mobile communication increases a company's competitiveness, as it helps achieve greater flexibility through the use of wireless communication to automation devices and industrial terminal devices. As a result you can simplify maintenance work, reduce service costs and downtimes and deploy your personnel optimally.

Industrial Wireless LAN (IWLAN) is based on the WLAN standard IEEE 802.11 and provides extensions that are particularly suited to demanding industrial applications with real-time and redundancy requirements. Customers are thus provided with a single wireless network for both process-critical data and uncritical communication.

REAL TIME CONTROL PROCESSES IN WPT-V2G SYSTEMS

Kaviya – IV Year

Information processes in V2G do not impose specific requirements for the data exchange in terms of the management of these processes in real time. For WPT with technology ICPT (Induction Coupled Power Transfer) “compensation” mechanism can be used. It is based on periodic measurement of certain parameters in the inductively coupled circuits for their further correction with the aim to maintain the efficiency of energy transfer.

Compensation for WPT is considered in patents by Gazulla et al. (2011) and paper by Saltanovs (2015). For an example in Dai et al. (2013) due to the load resistance changes with time the compensation process is introduced. Process is as follows: periodically measuring voltages and transfer these values in digital form wirelessly to the primary side of the system to adjust the generated pulses.

The purpose of this regulation is to maintain a constant voltage on the load. Time interval for the compensation cycle ΔTcycle is determined by external processes, as a result of which the circuit parameters are changed. For the objects that are charged in motion (like EV in WPT-V2G) one could estimate ΔTcycle for effective compensation as equal to several ms. Further, we estimate this time interval. Packets delay in wireless links

Wireless communication network for WPT-V2G the analysis leads to the conclusion that the architecture of information processes and data flows in WPT–V2G systems can be provided using wireless technology for data transfer. It may be necessary to build such network based on different wireless technologies

Along with the communications network serving information processes, which are not supposed real-time processes with restrictions for data packets delays, it may be necessary “technological” wireless network in which the packet delay will be strictly limited. The need of such network will be determined by the characteristics of the processes of compensation for ICPT when the wireless data link is used to transfer parameters of circuits between modules. Communication network architecture for WPT-V2G system. Informational (blue) and technological (red) networks. Vehicles (EV) may be in motion.