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Voice Connectivity penetration and declaration to rural area of Gujarat in mehsana district using WiMAX

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Abstract: - WiMAX as a broad-band access solution ahead of LTE and other is competing technologies due its long range and high bandwidth. Voice Over IP (VoIP) will potentially be the destroyer application for up-and-coming market like Gujarat rural area. In this Research we recommend a Kiosk based WiMAX infrastructure model to provide voice connectivity to rural area of Gujarat in mehsana district villages. In the proposed kiosk model, plain old telephones are connected to a WiMAX sub- scriber station using Foreign Exchange Subscriber(FES) and a Media Gateway. The innovation of the kiosk based infrastructure models is that it has low deployment cost from a service provider viewpoint, and almost negligible equipment cost for the end user. to make the kiosk based model cost-effectively sustainable. the number of simultaneous voice calls that can be supported over the WiMAX subscriber stations needs to be maximized. This research proposes a Dynamic Frame Profile algorithm to maximize the number of VoIP calls supported over a single subscriber station. A performance evaluation of the proposed Dynamic Frame Profile algorithm is also carried out to study its effectiveness and reported in this research.

Keywords:- PCO, Wimax, wifi, Kiosk based infrastructure model, VoIP,

I. INTRODUCTION

Connectivity is very important to any business and society so for up-and-coming markets like Gujarat. only half a percent of Indian population has residential Internet access, about 149.75 million [2].Gujarat had 21.8 million cellular subscribers and 322,656 internet/broadband subscribers in 2009 [1]. Telephone subscribers exist in India as of March 2006, which translates to approximately 1 telephone line per 1000 users. Out of the 149.75 million subscribers, about 49.75 million are fixed wire line telephone subscribers, and 100 million are mobile subscribers. Fixed telephone subscriber growth has had only marginal growth as compared to mobile subscribers due to the following two reasons. first, the cost of deployment is of fixed mobile telephones is about 27000 (Appro.) as compared to mobile telephones 4500 (Appro.) for this reason making it difficult for telecom operators to break even in terms of revenue. Secondly, the geographical and ground conditions in rural areas in India is not conducive to placing optical and copper cable, thus making is fixed wire- line deployment even more expensive. In order to close the gap of providing voice connectivity between the developed and developing nations, the Government of India came up with a scheme of Public Call Office (PCO). PCOs are small kiosks where a user can go and access telephone services. The PCOs/kiosk model also provide a cost effective way to provide voice connectivity especially in rural areas. We propose the use of WiMAX based network to resolution voice connectivity penetration to rural areas of mehsana district. Learning from the success of the PCO model. We extend this model in this research and propose a Kiosk based WiMAX infrastructure model for Rural Gujarat in mehsana district voice connectivity. We organize this research as following five parts.

Part 1 :- We motivate the require for the use of WiMAX to provide voice Connectivity in rural areas Of Gujarat in mehsana district.

Part 2 :- The Kiosk based infrastructure model which uses media gateway to provide voice connectivity.

Part 3 :- The technical challenges concerned in Kiosk based model.

Part 4 :- Enhancement to WiMAX called as Dynamic Frame Profile algorithm to increase VoIP capacity.

Part 5 :- A performance evaluation of Dynamic Frame Profile algorithm.

II. LITERATURE REVIEW

2.1 Use of WiMAX to provide voice Connectivity in rural areas of Gujarat.

The geographical area of Gujarat is divided into following class.
List of the cities in Gujarat
Class I: Population range (>= 100,000)
Class II: Population range (>=50,000 and <100,000)
Class III: Population range (>=20,000 and <50,000)
Following table gives class wise split of the populations of Gujarat [4].

Sr.No.	Name	Population	Class	SI. No.	Name	Population	Class
1	Ahmadabad	36,94,974	I	73	Navagam Ghed	39,500	III
2	Surat	27,02,304	I	74	Padra	39,205	III
3	Vadodara	14,11,228	I	75	Jasdan	39,046	III
4	Rajkot	10,03,015	I	76	Jambusar	38,778	III
5	Bhavnagar	5,17,708	I	77	Chhaya	38,526	III
6	Jamnagar	4,98,344	I	78	Dehgam	38,082	III
7	Junagarh	2,23,341	I	79	Memnagar	37,284	III
8	Nadiad	1,96,793	I	80	Thangadh	36,880	III
9	Gandhinagar	1,95,985	I	81	Khabhalia	36,479	III
10	Morvi	1,78,055	I	82	Vyara	36,226	III
11	Bharuch	1,67,117	I	83	Chaklasi	36,101	III
12	Navsari	1,62,250	I	84	Rajpipla	34,923	III
13	Porbandar	1,58,856	I	85	Kali	34,220	III
14	Veraval	1,58,032	I	86	Balasinor	33,705	III
15	Surendranagar	1,56,161	I	87	Dwarka	33,626	III
16	Anand	1,56,050	I	88	Lunawada	33,369	III
17	Gandhidham	1,51,693	I	89	Idar	32,805	III
18	Mehsana	1,41,453	I	90	Kodinar	32,610	III
19	Bhuj	1,36,429	I	91	Sanand	32,417	III
20	Godhara	1,31,172	I	92	Rajula	32,395	III
21	Palanpur	1,22,300	I	93	Radhanpur	32,191	III
22	Vejalpur	1,16,086	I	94	Umreth	32,191	III
23	Patan	1,13,749	I	95	Bagasara	31,796	III
24	Kalol	1,12,013	I	96	Vijapur	30,961	III
25	Ghatlodiya	1,09,467	I	97	Bavla	30,871	III
26	Jetpur-Navgadh	1,04,312	I	98	Mehmedabad	30,768	III
27	Botad	1,00,194	I	99	Gariandhar	30,526	III
28	Valsad	98,758	I	100	Ranavav	29,645	III
29	Gondal	97,506	I	101	Dhandhuka	29,572	III
30	Ankleshwar	96,325	I	102	Vallabh Vidyanagar	29,378	III
31	Dahod	95,957	I	103	Karamsad	28,955	III

32	Amreli	95,307	I	104	Un	28,820	III
33	Khambhat	93,194	I	105	Sarkhej-Okaf	28,808	III
34	Ranip	92,498	I	106	Joshipura	28,767	III
35	Deesa	83,382	I	107	Mansa	27,922	III
36	Dhoraji	80,811	I	108	Kalol	27,903	III
37	Mahuva	80,726	I	109	Manavadar	27,563	III
38	Savurkundla	73,774	I	110	Ramol	27,550	III
39	Visnagar	73,488	I	111	Salaya	26,875	III
40	Vapi	71,406	I	112	Gadhada	26,754	III
41	Dhrangadhra	70,663	I	113	Tarseli	26,706	III
42	Chandlodiya	68,417	I	114	Karjan	26,358	III
43	Anjar	68,343	I	115	Talaja	26,104	III
44	Wadhwan	63,424	I	116	Khedbrahma	25,506	III
45	Keshod	63,257	I	117	Bhachau	25,389	III
46	Dholka	61,569	I	118	Pardi	25,275	III
47	Kadi	60,026	I	119	Dakor	25,205	III
48	Sidhpur	58,194	I	120	Jhalod	25,095	III
49	Bilimora	57,564	I	121	Jafarabad	25,086	III
50	Borsad	56,548	I	122	Vadnagar	25,033	III
51	Himatnagar	56,464	II	123	Kalavad	24,858	III
52	Mangrol	56,320	II	124	Vastrapur	24,656	III
53	Chandkheda	55,504	II	125	Dungra	24,524	III
54	Upleta	55,438	II	126	Halvad	24,325	III
55	Dabhoi	54,952	II	127	Kheda	24,136	III
56	Modasa	54,135	II	128	Vapi	23,844	III
57	Vijalpur	53,913	II	129	Dhrol	23,628	III
58	Unjha	53,876	II	130	Chhapra Bhatha	23,415	III
59	Viramgam	53,094	II	131	Chhota Udaipur	23,211	III
60	Bardoli	51,946	II	132	Rapar	23,057	III
61	Palitana	51,944	II	133	Tharad	22,815	III
62	Una	51,261	II	134	Jamjodhpur	22,661	III
63	Petlad	51,147	II	135	Sonagarh	22,431	III
64	Sihor	46,960	III	136	Prantij	22,282	III
65	Halol	44,473	III	137	Dhanera	22,172	III
66	Jodhpur	44,388	III	138	Umbergaon	21,684	III
67	Kapadvanj	43,950	III	139	Chorvad	21,240	III
68	Thaltej	42,713	III	140	Motera	21,141	III
69	Mandvi	42,355	III	141	Abrama	20,967	III
70	Vastral	41,919	III	142	Lathi	20,966	III
71	Wankaner	40,191	III	143	Parvat	20,693	III
72	Limbdi	40,071	III	144	Kheralu	20,141	III

The majority of revenues and expected growth rates are now coming from traditionally underserved Class II and Class III regions and therefore the focus on rural connectivity has never been more intense. Some of the characteristics of Class II and Class III regions are as follows.

Low Literary: The average literacy rate is Class II and Class III regions is low, hence providing Broadband Internet access will have limited revenue earning. The major revenue earnings for operators will come from voice communications or providing telephone connectivity.

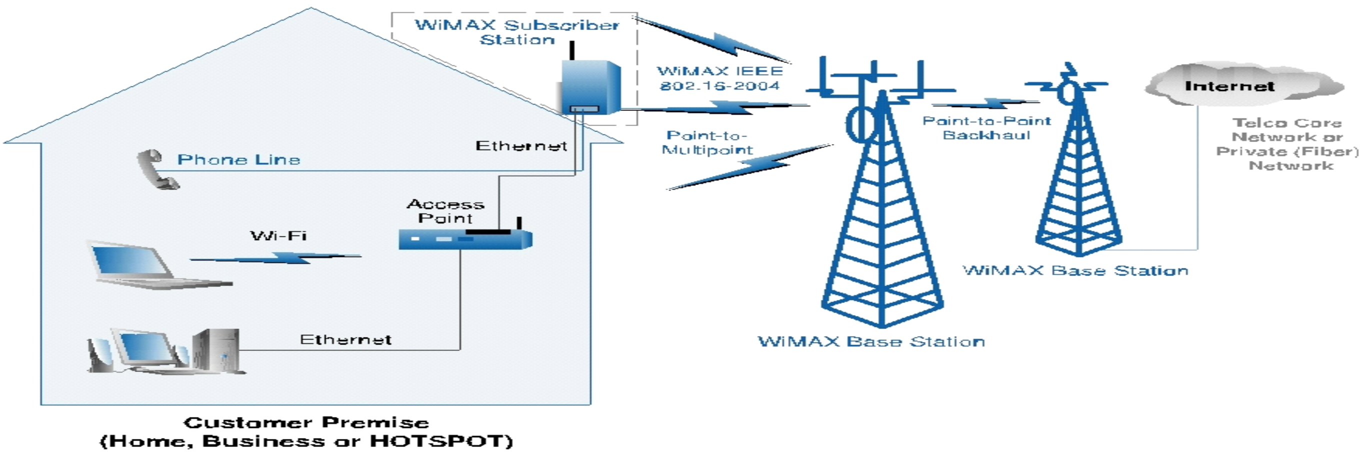
Low Income: As per the economic census 2011 report the per capita income in Class II region is about 13,000(Appro.) and in Class III region is 6,000(Appro.), thus providing low cost telecom services, especially voice is a challenge.

PC Penetration: PC penetration in Class II and Class III regions is on the increase still it is low due to low literacy and income. This again repeats the fact that providing voice connectivity will be the major focus for all telecom operators. Based on the above discussion it is clear that in addition to broadband access, voice communications is the primary driver for telecom operators in the rural areas. However, providing this rural voice connectivity has its own set of challenges.

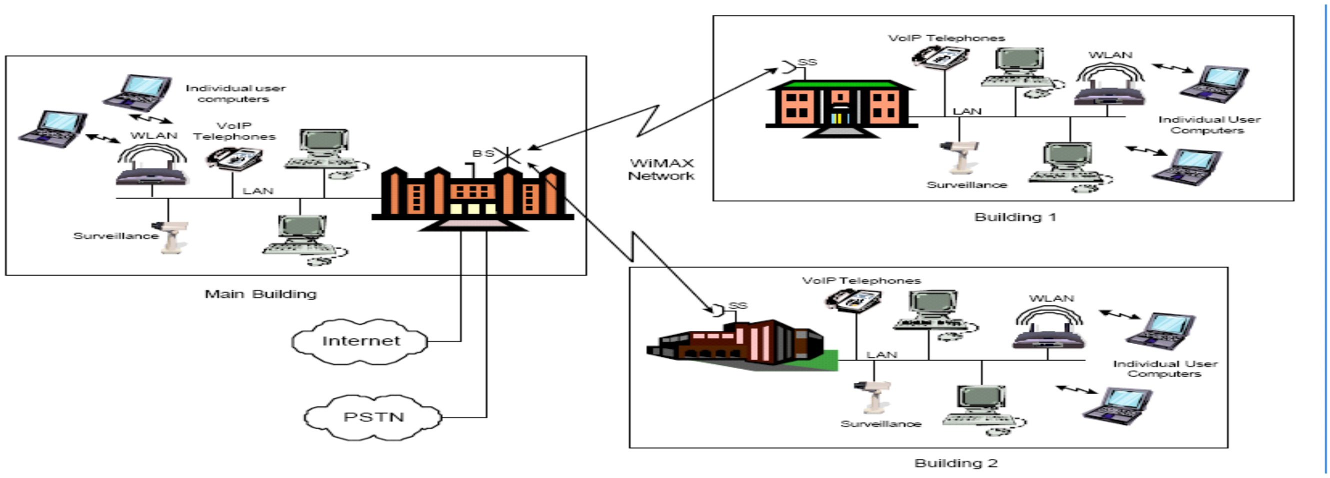
- ⇨ Telecom Regulatory Authority of India report determined to add an additional 7.5 million subscribers over the next 20 months in Class II and Class III regions, which translates to about 379,000 subscribers every month. Building a fixed wireline infrastructure for this is not an easy task and not economical [2].
- ⇨ The land conditions are not suitable for easy deployment of optical fiber or copper cables. Hence there is a need to look at use of wireless networks as backhaul networks.

As per above discuss it is clear that enabling voice communication takes precedence over other services. Considering the land conditions it is our idea that wireless networks is the right options to provide voice connectivity to rural areas, especially Gujarat villages. The other challenge is to reduce the deployment cost. One of the possible approaches to reduce the deployment cost is to use the kiosk model which we propose in this research. Also, given the fact that WiMAX is being used as an access technology, using WiMAX Customer Premise Equipment (CPE) is not an option for rural environment due to its high cost. Hence, the option for rural environment due to its high cost. Hence, the infrastructure must be compatible with existing Plain Old Telephone System (POTS). For this reason, I propose the use of a V5.2 Media Gateway. The following detail the planned infrastructure for providing voice connectivity.

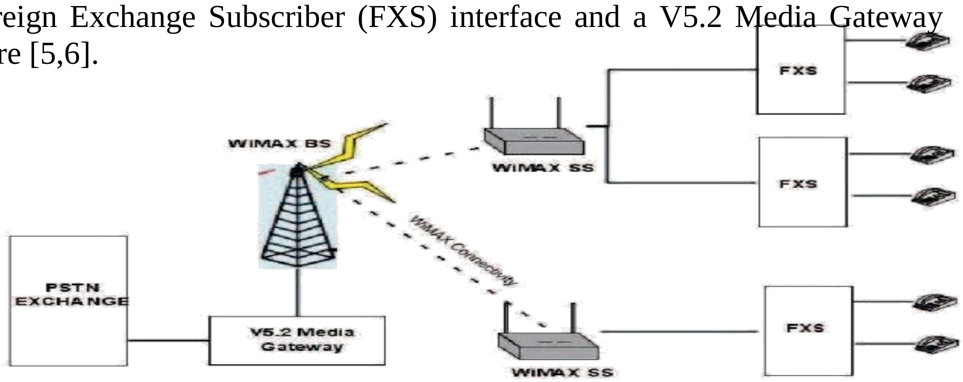
2.2 The Kiosk based infrastructure model which uses media gateway to provide voice Connectivity Rural kiosk model.



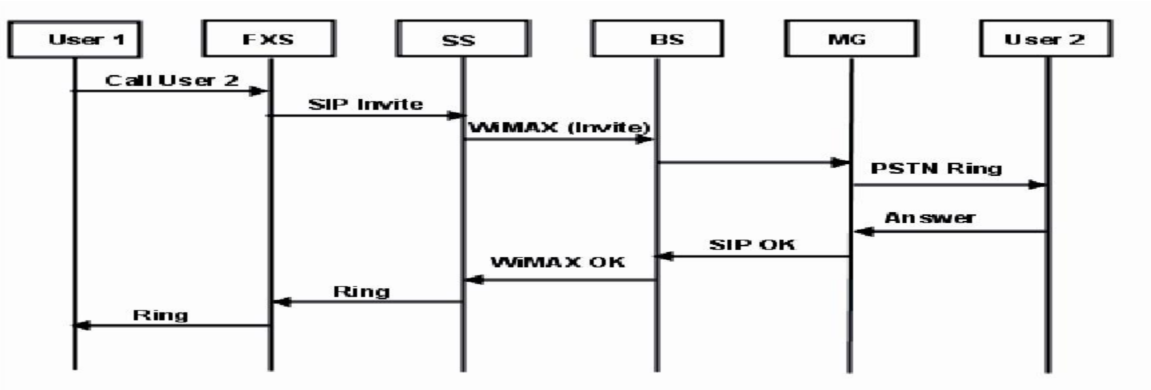
Above figure indicate a rural situation include of a few hundred households, or villages in Gujarat. Part 1 the aim is to provide POTS connectivity at an affordable cost to the end user as well as provide a cost effective deployment for the service provide. For this reason, we propose the use of a kiosk model as shown in following figure.



Each kiosk is a WiMAX Subscriber Station (SS), and provides voice service to set of houses (POTS). Each village will be served by a single WiMAX Base Station (BS) or alternately each WiMAX BS sector will service a village. To enable the POTS to be connected to the SS at the kiosk. We suggest the use of Foreign Exchange Subscriber (FXS) interface and a V5.2 Media Gateway as shown in following figure [5,6].



The FXS is placed at the Kiosk along with the SS. All the POTS serviced by the SS are terminated at the FXS. The FXS is responsible to convert the analog telephone call to a VoIP call. The Media Gateway which is based on the Media Gateway Control Protocol (MGCP) is responsible to provide the signaling support and interface with the PSTN network. In the proposed architecture is designed for fixed line connectivity however in the future the architecture can be extended to support mobile connections. A typical abstract view of the call flow for the proposed architecture is given in following Figure.



Above figure observe that the end user (User 1) is trying to make a PSTN call to User 2. The FXS is responsible to convert the analog or DTMF signaling to an VoIP message, which in this case is a

Session Initiation Protocol (SIP) invite. This SIP invite is received by the SS, and forwarded to the BS, which in turn forwards to the receiving SS over a WiMAX frame. An Enum server is used at the SS to convert the PSTN number to an equivalent SIP address. Once the receiver (User 2) receives the ring, the loop back signal is transmitted back as a SIP OK message to the caller (User 1). Detailed messaging is not discussed as a part of this research, as it is out of scope of this research. It is our faith that the above proposed model is not only going to be cost effective and practical for deployment for both the service provider and the end user.

2.3 The technical challenges concerned in Kiosk based model.

As per discuss part 2 one of the possible architectures for providing voice connectivity in rural India using WiMAX. This part focuses on the technical challenges involved in deploying the proposed infrastructure. WiMAX, which is based on 802.16[7] air interface standards. In the present architecture I assume the use of 802.16e[8] standard and 802.16e supports five classes of traffic, out of which two, viz. Unsolicited Grant Service (UGS) and Extended Real-Time Polling Service (ERTPS) are used for VoIP traffic From a service provider perspective, the challenge is to maximize the simultaneous number of users supported by the BS. Since the communication between two subscriber stations is VoIP based, the problem statement reduces to maximizing the VoIP capacity of the WiMAX network by enhancing the simultaneous VoIP calls supported by the SS. Our experiments indicate that a maximum of 10-12 simultaneous VoIP calls can be supported over a single subscriber station [9] without degradation in Quality of Service (QoS). In order to maximize the the number of VoIP calls I propose a Dynamic Frame Profile (DFP) algorithm. The algorithm can be implemented without having any extra overhead of messaging and can be incorporated in the current MAC architecture as recommended by 802.16 standard without any modifications [7,8]. The *PHY Synchronization Field* in DL-MAP sub-frame is used by the BS to inform the SS about the current frame length. The *PHY Synchronization Field* has a frame length field defined with codes from 0x00 to 0xff which represent the frame lengths respectively in msec. The first 3 fields are reserved for frame lengths of 0.5msec,1msec and 2msec. The rest of the fields are left for the service provider to define and we propose the use of these fields to convey the frame duration. For the sake of completeness. We briefly describe some of the related work in the following part.

III. ENHANCEMENT TO WIMAX CALLED AS DFP ALGORITHM

3.1 Enhancement to WiMAX called as Dynamic Frame Profile algorithm to increase VoIP capacity.

The basis of this algorithm is to dynamically change the frame duration based on VoIP load per SS. Let T_f denote the default frame duration. Let *Ts* denote the OFDM symbol duration, and hence the total number of symbols S_t per frame is given by Equation 1.

$$S_t = \frac{T_f}{Ts} \text{ ----------------------------- } (1)$$

bs denote the bandwidth allocated per symbol so the total bandwidth BF per frame is given by following Equation 2.

$$B_f = bs * S_t \text{ ----------------------------- } (2)$$

α_{ugs} and denote α_{ert} denote the fraction of the bandwidth reserved for UGS and ertPS class of traffic, then the total bandwidth reserved for VoIP Bvoip traffic is given by Equation 3.

$$Bvoip = \alpha_{ugs} * Bf + \alpha_{ert} * Bf \text{ -------------------- } (3)$$

If Bi denotes the bandwidth requested by ith VoIP session, the average bandwidth for VoIP session is given by Equation 4.

$$Bavg = \frac{\sum_{i=1}^{i=k} Bi}{k} \qquad (4)$$

where k denotes the total number of active VoIP sessions. The average number of Nvoip VoIP sessions supported by a single SS is given by Equation 5.

$$Nvoip = \frac{Bvoip}{Bavg} \qquad (5)$$

Nvoip needs to be maximized in emerging markets. The DFP algorithm proposed in this research monitors the VoIP call blocking probability Pb. the calls are blocked when the requested bandwidth Bi for a service flow cannot be allocated, based on the call admission policy. From Equation 1 and Equation 4, it is clear that Nvoip is maximized by increasing St, which implies changing the frame duration Tf . Let β denote the frame scaling factor, and Bjrej denote the bandwidth requested by the jth VoIP session that has not been admitted into the network. The proposed DFP algorithm is as following.

Set Frame Length = Tf , $Brej$ = 0, $Pprev$ = 0
if ($Ti \leq Th$)
On Arrival i^{th} VoIP session
If ($Bi + Bins \leq Bvoip$)
Admit Session i
Update call admission Count
if $li \leq lth$
    Recompute frame length
Endif
  else
    Reject Session
Update Call drop count
    $Brej = Brej + Bi$
  Endelse
Endif
else
Compute instantaneous Pi
if ($Pi = 0$), set $Pi = Pprev$
Endif
$Pb = W1 \times Pprev + w2 \times Pi$,
where $0 < w1 \leq w2 < 1$
$Pprev = Pb$
If $Pb \geq Pth$
$T_f^{new} = Tf + \beta \times = \frac{Brej \times Ts}{bs}$
Set $Tf = T_f^{new}$
Endif
Endelse
Endif

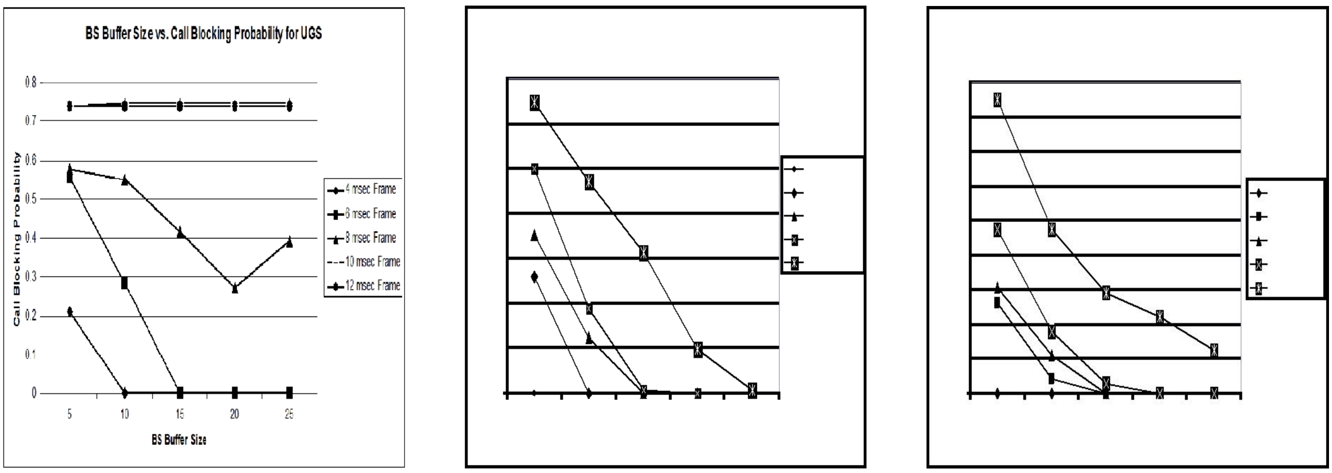
where *Bins* and *Ti* denote the immediate bandwidth and time respectively. The call drop probability is calculated over periodic time intervals of *Th*. immediate call drop probability does not give an accurate reflection of the current state of the network and decisions based on the immediate call drop probability lead to frequent changes in frame length, leading to degradation in QOS as the DFP algorithm does not converge. In DFP algorithm, the call drop probability is calculated over a window of time. Let *Pprev* denote the call drop probability at the *t*- time instant and *Pb* denote the over all call drop probability at time instant *t*. While calculating *Pb*, appropriate weight factors are used. Steps 3-9 in above algorithm compute the number of accepted and rejected calls in the current time interval and the total bandwidth required for the rejected sessions. On admission of a call if the load per SS exceeds a threshold load per SS *lth*, a new frame duration is calculated. On completion of the current time interval the immediate call drop probability is calculated. If no calls arrive in the current time duration or no calls are dropped the immediate call drop probability Pi is zero. In such a case the immediate call drop probability is re-initialized to the previous call drop probability Pprev. This give the algorithm to maintain the current state of frame duration for a longer time interval, thus stabilizing it. Step 16 in above algorithm calculates the weighted average call drop probability with weights 0 < w1 ≤ w2 < 1. Once the call blocking probability exceeds the threshold call blocking probability Pth, a new frame duration T f new is computed in step 19 using Equations 1 and Equation 2. The following section reports the performance evaluation of the DFP algorithm.

3.2 A performance evaluation of Dynamic Frame Profile algorithm.

A performance evaluation using Network Simulator (NS-2) [10]. patched with the WiMAX extension proposed in is carried out to study the effectiveness of the proposed DFP algorithm[11]. Simulations are approved out for three different state.

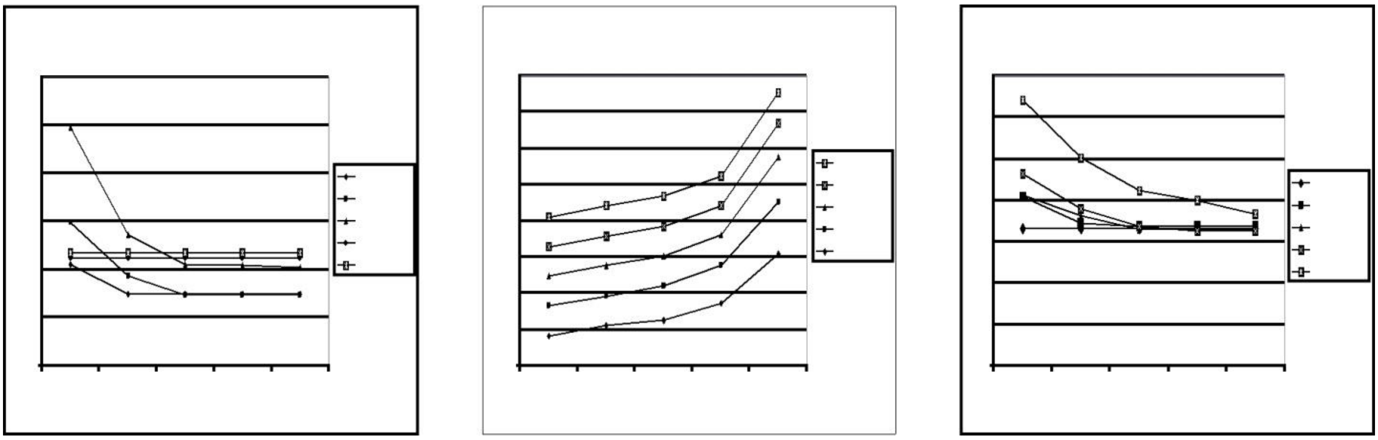
1) Voice carried only over UGS.
2) Voice carried only over ertPS.
3) Voice carried over UGS + ertPS

The simulation state include of a single BS(Base Station) and 8 SS (Sub Station). The call duration is Poisson distributed with an average of 16 seconds. The inter arrival time between calls is exponentially distributed with an average time of 0.1 seconds. Such high loads of approximately 64 connections were considered to get practical simulations results for a rural state. The results are averaged over multiple runs of different frame durations as followings.



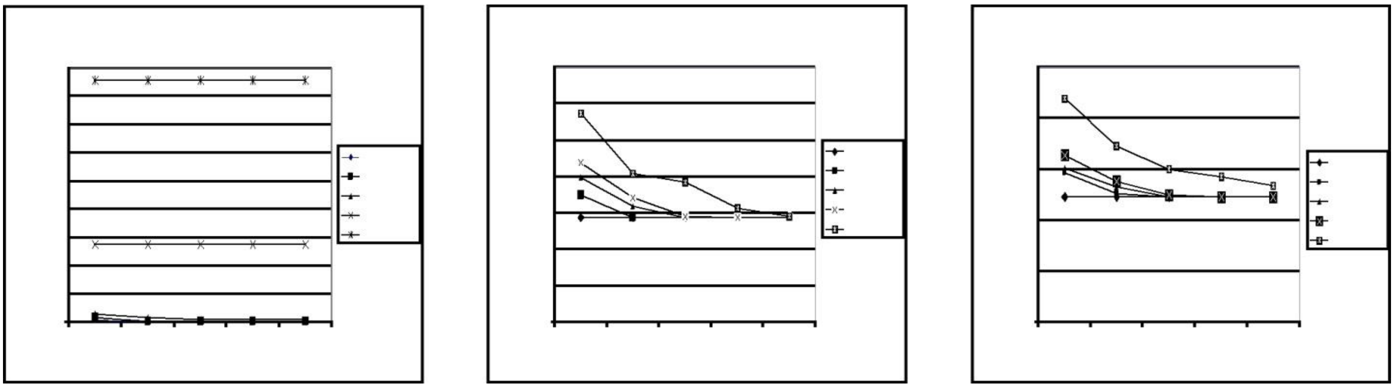
(a) UGS (b) ertPS (c) UGS + ertPS

Figure 1: Call Blocking Probability for UGS, ertPS, and UGS + ertPS



(a) UGS (b) ertPS (c) UGS + ertPS

Figure 2: Delay Jitter for UGS, ertPS, and UGS + ertPS



(a) UGS (b) ertPS (c) UGS + ertPS

Figure 3: Packet Loss Rate for UGS, ertPS, and UGS + ertPS

In Above figure the figure 1 gives the call blocking probability for the three states for different buffer size *(in terms of packets)* at the Base Station. Figure 1 conclude that a BS buffer size of 25, the call blocking probability tends to zero, indicating that by varying the frame duration, the number of simultaneous VoIP calls that can be accommodated increases. Another important metric for VoIP based applications is the delay jitter and packet loss probability. When, the number of VoIP calls increase, the delay jitter should be at most 50 msec and the packet loss probability should be within negotiated QoS bounds. Figure 2 gives the delay jitter for the three states. From Figure 2(b) observe that with increase in buffer length, the delay jitter increases. Note that in all the three states, the delay jitter is well within the tolerable limits of 50 msec. Hence, varying the frame length does not impact the quality of the VoIP calls. Figure 3 plots the packet loss probability for the three states as following.

1) Figure 3(a), examine that the packet loss rate increases with increase in frame duration and reduces with increase in buffer size at the BS.
2) Figure 3, examine that the packet loss rate increases with increase in frame duration and reduces with increase in buffer size at the BS.

3) In the present picture as the frame duration increase beyond 8 msec, the packet loss rate increases beyond the threshold value. Hence, for the present scenario the frame duration cannot be increased beyond 8 msec.

IV. CONCLUSION

We have proposed In this research a Kiosk based infrastructure model to provide voice connectivity in rural area of Gujarat in mehsana district. The outstanding features of the Kiosk based infrastructure model take in relatively low use cost and cost-effectively for the end user. The key to the success of the proposed Kiosk model is to maximize the number of voice calls supported over a single WiMAX subscriber station. This research proposes and evaluates a Dynamic Frame Profile algorithm. It is my principle that using the Kiosk model, it will be feasible to provide voice connectivity to rural area of Gujarat in mehsana district in a cost effective way.

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