Wireless LAN | VoIP | QoS

Technology Reports

Advanced Wireless LAN VoIP Technology

A technical overview is given of an optimal access point selection method and an autonomous distributed scheduling MAC method that take QoS into account. Those methods are proposed as new technology to improve the communication quality of wireless LAN VoIP. Even when the number of simultaneous calls in the same area increases to about 50% above the maximum number of calls possible with the conventional method, the proposed methods allow the same voice quality as before the increase.

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1. Introduction

The use of Institute of Electrical and Electronics Engineers 802.11 (IEEE 802.11)*1 wireless LANs is rapidly spreading as the price of hardware decreases and various networks expand in businesses, homes and public spaces. Computers and other such data terminal products that have a wireless LAN interface are also rapidly spreading in the market. In recent years, application of wireless LANs to real-time applications typified by Voice over IP (VoIP) as well as data communication has been anticipated.

In a wireless LAN, multiple independent systems share an unlicensed frequency band known as the Industrial, Science, and Medical (ISM) band, so a means of ensuring communication quality when a wireless LAN is applied to VoIP is essential. However, with the general access point (AP) selection method that considers only the Signal to Noise Ratio (SNR)*2 and the number of stations connected, or the Enhanced Distributed Coordination Access (EDCA) which is Quality of Service (QoS)*3 control method defined by IEEE 802.11e*4 [1] alone, it is generally difficult to maintain good voice quality when the number of simultaneous calls increases.

In this article, we provide an overview of the technology we devised to solve those issues.

2. Overview of Wireless LAN VoIP

The general configuration of a wireless LAN VoIP system for an office or other such location is shown in **Figure 1**.

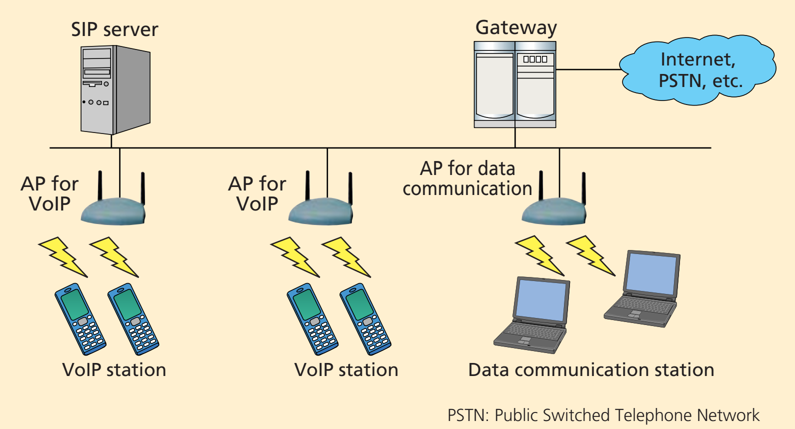

Figure 1 General configuration of a wireless LAN VoIP system for offices, etc.

*1 **IEEE 802.11**: An international standard for wireless LAN defined by IEEE of the United States.

*2 **SNR**: The ratio of desired signal power to interference signal power in radio communication.

*3 **QoS**: A level of quality on the network that can be set for each sevice. The amount of delay or packet loss is controlled by controlling the bandwidth that the service can use.

*4 **IEEE 802.11e**: International standard for wireless LAN QoS (see *3) technology defined by IEEE.

A wireless LAN VoIP system comprises "VoIP stations," "APs," a "Session Initiation Protocol (SIP) server[*5]," a "gateway" and other components. To secure a wide coverage area, multiple APs are generally deployed within an office.

On the other hand, the number of frequency channels that can be used for a wireless LAN is limited, so there is a possibility of mutual interference with other systems (data communication, etc.) that use the same frequency channel as VoIP communication. For that reason, when there are multiple APs in a single wireless LAN VoIP communication area, the VoIP stations must take into account factors such as the amount of traffic that the AP is processing and the number of stations connected to it rather than simply selecting the nearest AP.

Also, efficient packet transmission in the Medium Access Control (MAC) layer[*6] is needed to prevent voice packet collision as the number of calls increases. EDCA, the method for QoS control in a wireless LAN, implements priority control by classifying applications into four Access Categories (ACs), voice communication, video communication, data, and background, according to the level of priority. However, when the number of voice stations that have the same priority increases, the probability of packet collision also increases, so EDCA alone cannot be expected to achieve highly precise QoS control [2].

In this article, we propose an optimal AP selection method that takes QoS into account and is implemented by adding functions to the AP and VoIP station, and an autonomous distributed scheduling MAC method that allows more simultaneous calls through changes only by adding functions on the VoIP station side. We show that the proposed methods can maintain voice call quality even when the number of simultaneous wireless LAN VoIP calls increases.

3. Optimal AP Selection Method Considering QoS

3.1 Purposes

In a wireless LAN , the quality of voice calls deteriorates greatly due to packet collision when the number of simultaneous calls increases. As a result, optimal selection of AP is indispensable to the improvement of the overall performance of voice applications in an area where multiple Basic Service Set (BSS)[*7] are available for access. In particular, AP selection that takes QoS into account is desired in the case that real-time communication such as VoIP co-exists with non-real-time data communication.

Previous studies have presented different AP selection criteria according to the amount of traffic and available bandwidth, etc. [3]-[5]. Those approaches, however, do not take into account the QoS differentiation and the hidden terminal problem[*8], which greatly increases packet collision probability [6]. As the amount of traffic varies among the ACs which decide the access priority in EDCA, the VoIP calls are differently affected by the respective AC traffic. It is therefore necessary to emphasize the amount of traffic that corresponds to the ACs with the same as and higher priority than VoIP.

In this article, we propose an AP selection method that considers four factors, which are "the supportable physical layer data transmission rate," "the number of connected stations per AC," "the effect of hidden terminals," and "the effects from different ACs."

3.2 Proposed Method

The sequence for when the proposed method is applied to passive scanning[*9] is shown in **Figure 2**, where AP1 and AP2 are assumed to use non-overlapping channels (f_1 and f_2).

In the proposed method, AP selection with QoS awareness is achieved by including the number of stations corresponding to each AC in the beacon[*10]. The VoIP station obtains from the beacon the current load and number of connected stations for each AC inside the BSS where it is currently scanning. In the same way, it switches to other available channels, checking the APs that are accessible based on the presence of beacon, and then determines the number of connected stations for each neighboring AC.

*5 **SIP server**: A server that performs call connection control in VoIP communication.

*6 **MAC layer**: A layer that has a control function for preventing packet collisions when sharing communication lines among multiple nodes. This layer is a sublayer of the data link layer in the OSI 7-layer model.

*7 **BSS**: A unit of wireless LAN configuration that comprises an access point and multiple stations.

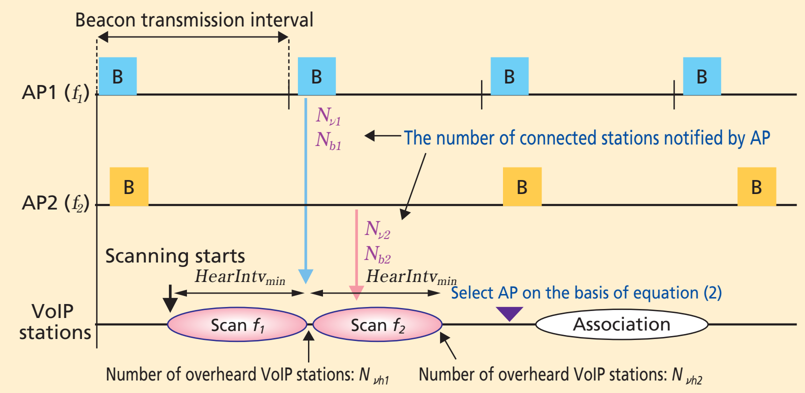

Figure 2 Sequence of the optimal AP selection method considering QoS

We denote the number of VoIP stations overheard by the VoIP station as $N_{\nu hi}$, the number of overheard data communication stations as N_{bhi}, the number of VoIP stations obtained from the beacon as $N_{\nu i}$ and the number of data communication stations as N_{bi}, and the currently scanned channel index as i. In addition, the VoIP station uses the SNR of the received beacon to determine the maximum supportable data rate ν_i. The VoIP station switches to the next frequency channel after at least one beacon is correctly received and a pre-set observation time-out time, $HearIntv_{min}$, has expired.

To serve as the metric for AP selection applied to VoIP stations, $f_{\nu i}$ is defined as in equation (1) [7].

$$f_{\nu i} = (N_{\nu i} - N_{\nu hi}) \cdot L/\nu_i \qquad (1)$$

Here, L is the mean data packet length.

The number of hidden terminals is the number of stations whose signal cannot be received by the counterpart, so $(N_{\nu i} - N_{\nu hi})$ shows an estimate of the number of hidden VoIP station. Furthermore, L/ν_i denotes the average required time for data transmission.

As the quality of service is affected by the transmission whose priority is the same or higher than that of the AC to which a station belongs, Eq. (1) is generalized as Eq. (2) when all of the ACs are considered.

$$f_{mi} = \sum_{k=1}^{m} (N_{ki} - N_{khi}) \cdot L/\nu_i \qquad (2)$$

Wherein, k is defined as a value of 1, 2, 3 or 4, with k=1 indicating the index of the highest-prioritized AC (voice communication in EDCA).

Here, we explain the operation for the case in which the proposed method is applied to passive scanning. Similarly this method can also be used for active scanning[*11] by inserting information such as $N_{\nu i}$ and N_{bi} into the probe response frame[*12].

3.3 Frame Format

Up to now many load metrics have been included in beacon or probe response frames to provide reference for association. However, none of the elements differentiates the load by access priority, which is necessary for selecting the optimal AP for VoIP communication. Hence we define a new Information Element (IE)[*13], AC station count, to describe the number of connected stations corresponding to each AC (**Figure 3**) and propose inserting it into beacon or probe response frames. This IE includes the following new information.

- Station Count Bitmask:
  Shows the ACs that have the Sta-

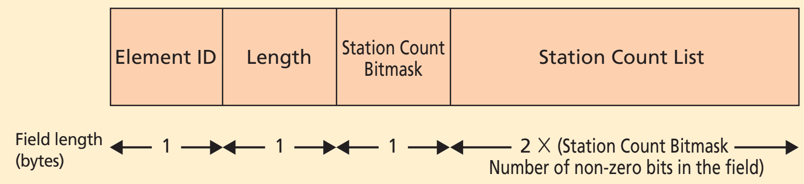

Figure 3 New IE in the proposed method

*8 **Hidden terminal problem**: Terminals located in areas that cannot receive signals of each other and cannot recognize other's communication status. A phenomenon by which packets submitted simultaneously by hidden terminals collide and call quality degrades is called the "hidden terminal problem."

*9 **Passive scan**: A method of discovering access points in which the station receives the beacon (see*10) that is periodically transmitted by access points.

*10 **Beacon**: Common information that is transmitted by access points periodically in intervals of from tens of ms to hundreds of ms.

*11 **Active scan**: A method of discovering access points in which the station transmits a probe request packet.

tion Count specified in the following Station Count List. The AC and Bitmask mapping is shown in **Table 1**.

- Station Count List:
  Shows the number of connected stations corresponding respectively to the non-zero bits in the Station Count Bitmask field.

This proposed frame format has been adopted by the draft of IEEE 802.11v[*14] [8][9].

3.4 Performance Evaluation

We evaluated the proposed AP selection method in terms of packet loss rate[*15] and the number of reassociations[*16] by computer simulation. Two IEEE802.11b[*17] BSSs work on different channels and provide overlapped coverage (**Figure 4**). The VoIP stations are placed randomly, with 80% in the AP1 area and 20% in the AP2 area. Four data communication stations were also placed to serve as interference for VoIP communication. The physical layer transmission rate was determined according to the SNR of the recieved beacon [10]. The three methods we compared were to select the AP with the highest SNR (Max. SNR), select the AP with the fewest connected stations (Min. Nmt) and the proposed method. In this simulation, the packet loss rate of 0.1 was used as a reference value to trigger reassociation for voice quality.

The simulation results for the uplink packet loss rate with respect to the total number of stations and the number of VoIP stations in the area are shown in **Figure 5**. We can see that the proposed method allows up to about 40% more simultaneous calls than the conventional method.

The simulation results for the number of associations versus the number of VoIP stations in the area are shown in **Figure 6**. The proposed method can reduce the number of reassociations by about 25% to 50%. Therefore, the proposed method avoids frequent switching between neighboring APs and further saves power on AP reassociation.

Table 1 Station count bitmask field

Bit	AC
0	AC_BE
1	AC_BK
2	AC_VI
3	AC_VO
4～7	Reserved

AC_BE: Data
AC_BK: Background
AC_VI: Video signal
AC_CO: Voice signal

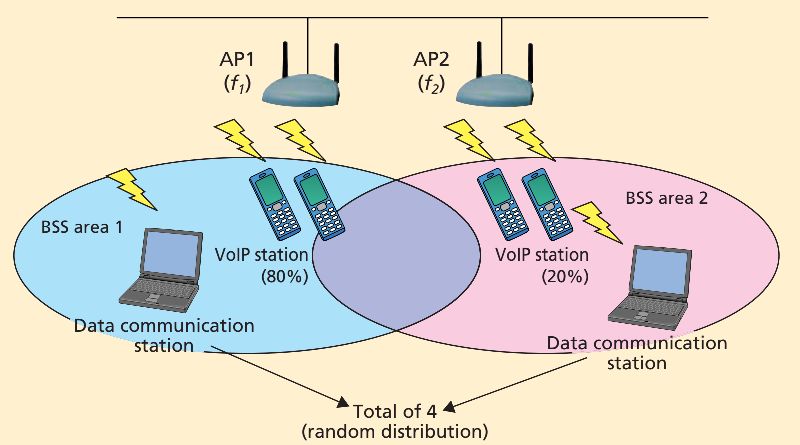

Figure 4 Network configuration used in the simulation (proposed AP selection method)

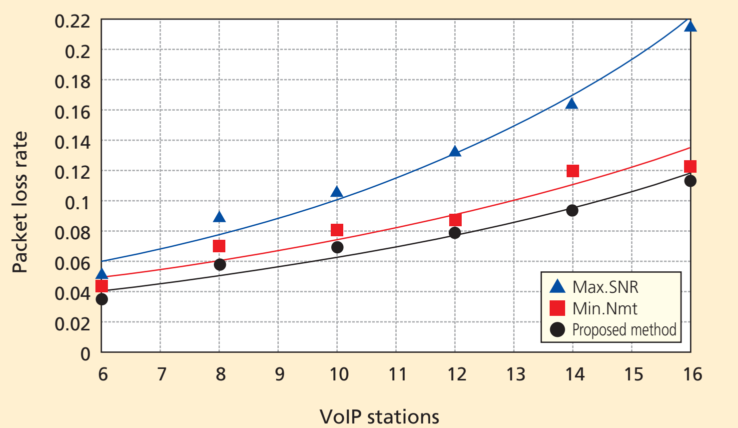

Figure 5 Results of packet loss rate (proposed AP selection method)

*12 **Probe response frame**: The packet that a station receives from an access point in response to a transmitted probe request packet. Receiving the probe response frame informs the station of the location of the access point and the available transfer rate.

*13 **IE**: An information element for the number of stations connected, the available transfer rate and other such information. It is contained in a packet in the beacon or probe request/response packet, etc.

*14 **IEEE 802.11v**: An international standard concerning extension of the wireless LAN wireless cell management method.

*15 **Packet loss rate**: The proportion of the total number of packets transmitted to the packes that do not arrive normally because of interference, packet collision, etc. in a wireless cell.

4. Autonomous Distributed Scheduling MAC Method

4.1 Purpose

EDCA, the QoS control method used in wireless LANs, implements priority control by classifying applications into four ACs according to the level of priority and giving each AC a different packet transmission start time. An overview of an EDCA access scheme is shown in **Figure 7**. A high-priority AC has fewer time slots before packet transmission than a low-priority AC, with the result that it has many transmission opportunities. Here, the number of time slots before packet transmission is an Arbitration Inter Frame Space (AIFS) or random backoff time (Backoff) parameter assigned to each AC according to its priority. Low-priority AC, on the other hand, has more time slots before packet transmission. By assigning differences in the number of time slots before packet transmission according to the application priority in this way, higher transmission priority can be given to VoIP stations, even when VoIP stations and data communication stations co-exist in the same frequency channel. Nevertheless, EDCA can apply priority control only between different AC, and when there are many VoIP stations in the same area, proper operation of priority control is not possible.

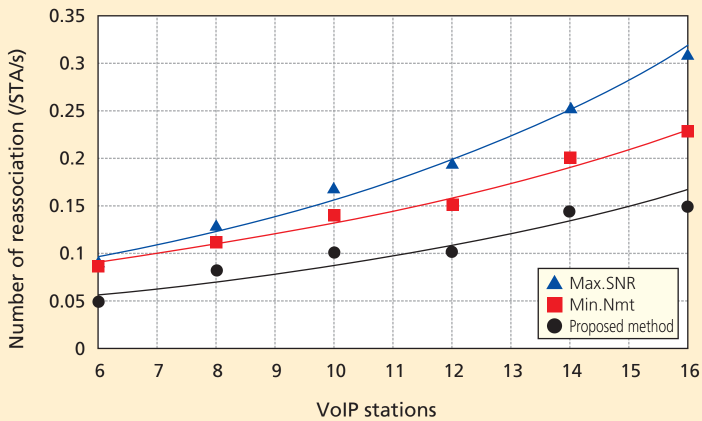

Figure 6 Results of number of reassociation (proposed AP selection method)

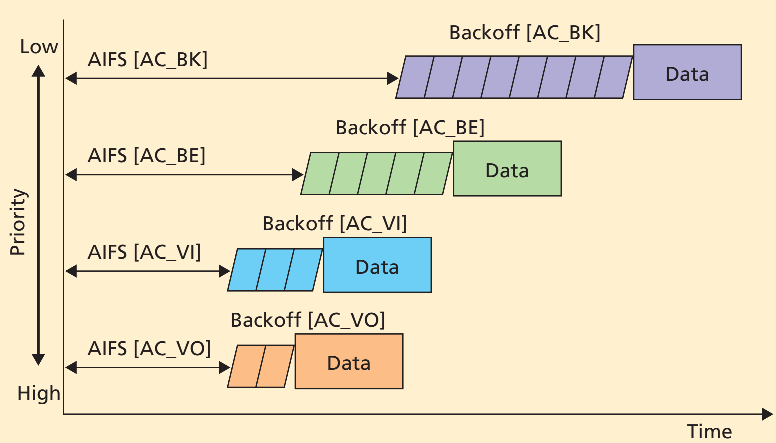

Figure 7 Overview of the EDCA access method

In addition to EDCA, IEEE 802.11e also specifies a QoS control method by centralized control scheme, which is called Hybrid coordination function Controlled Channel Access (HCCA), but HCCA is known to have many issues, including polling collision in areas of cell overlap [2][11].

In view of those issues, we propose in this article an MAC protocol for autonomously setting the transmission order and transmission time among VoIP stations, thus achieving high-quality wireless LAN VoIP communication. We considered the following three points as design guidelines for high implementability.

- Can be implemented at the terminal station alone, which means that existing AP can be reused for convenience to users
- Can be implemented in software alone to suppress hardware implementation impact
- Backward compatibility to preserve communication with and between

*16 **Reassociation**: A call establishment procedure that is executed before communication between a wireless LAN station and access point begins and when the call is disconnected.

*17 **IEEE 802.11b**: A wireless standard defined by IEEE. Uses the 2.4-GHz frequency band and supports a transfer speed of 11 Mbit/s. Upward compatible with the 54 Mbit/s 802.11g standard.

existing stations

4.2 Proposed Method

The VoIP station flow chart for the proposed method is shown in **Figure 8**. First, each VoIP station periodically counts for downlink packets transmitted by the AP to determine for whether or not there are calls in other VoIP stations within the same BSS, reads MAC address from the destination address field of the MAC header, and creates a list of the stations within the cell, such as shown in **Table 2**. Because all of the stations in the same BSS can receive the downlink packets, a list of the stations in the cell can be shared without defining a new sequence or control packet, etc. As one example, the MAC addresses in the station list are in increasing order. For example, if the station's MAC address in Table 2 is "xx:xx:xx:03:02:06", that station recognizes that, of all the stations that are in the cell within the specified period, it can be the second to transmit.

Each station sends and receives voice packets in the order established by the above procedure. Also, in the case of successive sending and receiving failures due to a new station entering after the schedule has been established or other such reason, the VoIP station sends a re-synchronization request. The re-synchronization request is sent to all of the stations in the BSS by a broadcast transmission that is relayed by the AP. The stations that receive the re-synchronization request once again begin to monitor downlink packets to establish a schedule. During the schedule establishment, calls are sent and received according to the conventional EDCA method.

After the schedule has been set according to the procedure of Section 4.1, each VoIP station performs adaptive control on the AIFS Number (AIFSN), CW_{min}[*18], CW_{max}[*19] or other such EDCA parameter according to the set schedule, and then sends and receives, averting packet collision by setting the call priority period. The transmission priorities set by the proposed method and the conventional EDCA method are respectively shown by the solid and broken lines in **Figure 9**. The transmission priority corresponds to the inverse of the AIFSN or CW_{min}. With the conventional method, the transmission priority, which is to say the EDCA parameter, is always assigned statically. In the proposed method, on the other hand, the VoIP station changes AIFSN, CW_{min}, and CW_{max} dynamically at the time of transmission as determined by autonomous distributed packet scheduling within the specified voice codec period, and

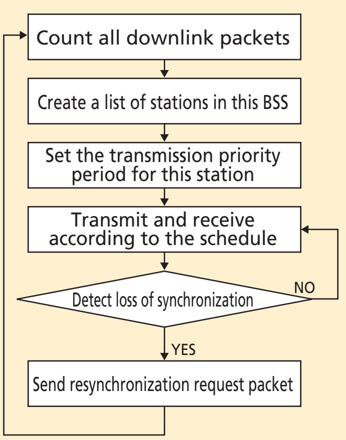

Figure 8 Flowchart for autonomous distributed scheduling MAC method

Table 2 Example of a station list

Number	MAC address
1	xx:xx:xx:01:01:03
2	xx:xx:xx:03:02:06
3	xx:xx:xx:07:09:02
…	…
…	…

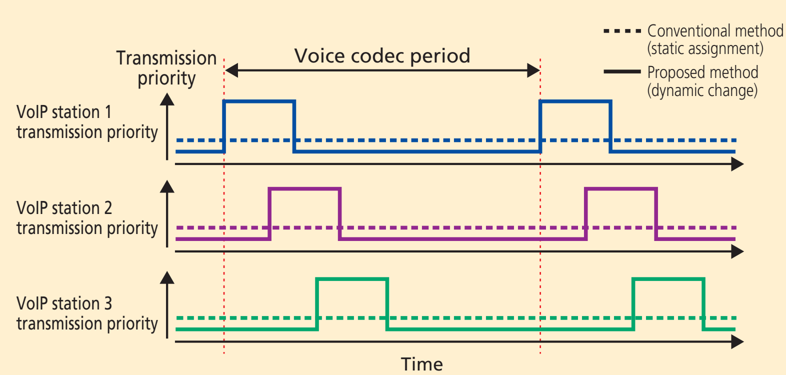

Figure 9 Allocation of transmission priority periods in the proposed method

*18 **CW_{min}**: A parameter that sets the maximum value of the random backoff time before packet transmission when wireless LAN packets are sent the first time.

reduces the number of time slots before packet transmission relative to the other stations only for a specific period. This period is called the call priority period. The interval for shifting the call priority period between stations can be set, for example, to the time required for uplink and downlink sending and receiving. That value can be computed by using voice codec and transfer rate information. In the proposed method, the packet transmission timing for the stations can be distributed, as shown in Fig. 9, so improvement of packet collisions can be expected.

The method does not only allocate transmission opportunities to stations that implement the proposed technology. Because the autonomous distributed obtaining of right to transmit is performed by Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), this method does not interfere with transmission by stations that implement only existing technology. In particular, schedule failure due to transmission interruption of stations that do not implement the proposed technology can also be controlled by making a series of call priority periods as contiguous as possible and setting a longer period that is not a priority period for any of the stations that implement this technology. Furthermore, this method is highly compatible with the Unscheduled-Automatic Power Save Delivery (U-APSD) power save system specified by IEEE 802.11e, and allows highly efficient intermittent reception.

Because this method does not require a new sequence, it can be implemented simply by changing the station software. It also does not require a change in the AP, so it satisfies all of the design conditions described in Section 4.1.

4.3 Performance Evaluation

We evaluated the proposed MAC method when used with IEEE 802.11b by computer simulation.

First, we simulated the case in which multiple VoIP stations begin communication simultaneously with no established schedule. The results for the transient change in the packet loss rate are shown in **Figure 10**. The proposed method converges to a packet loss of about zero in about 100 ms. The explanation for that result is that no schedule has been established immediately after the beginning of the simultaneous communication, so the various stations execute packet transmission with arbitrary timing by the conventional EDCA method and packet loss occurs. After that, each station sends and receives in the order of the station list according to the scheduling performed among the stations.

The simulation results for packet loss rate and throughput when the number of stations is increased are shown in **Figure 11**. With EDCA, the packet loss rate increases as the number of simultaneous calls increases, and the throughput decreases at the same time. This result signifies that the conventional method cannot guarantee the theoretical value for the maximum number of calls [12]. With the proposed method, on the other hand, there is no increase in packet loss rate or decrease in throughput, even when the number of simultaneous calls increases by about 50%. In other words, this shows that the proposed method can greatly reduce the packet collision probability that accom-

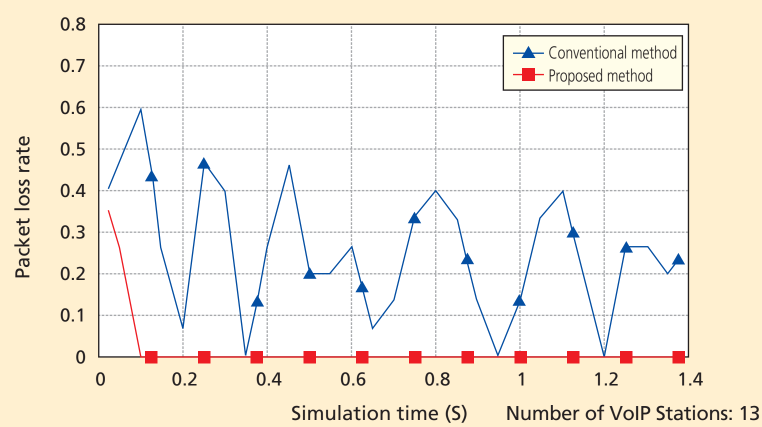

Figure 10 Simulation results for scheduling convergence time

*19 **CW_{max}**: A parameter that sets the maximum value of the random backoff time before packet transmission when wireless LAN packets are resent.

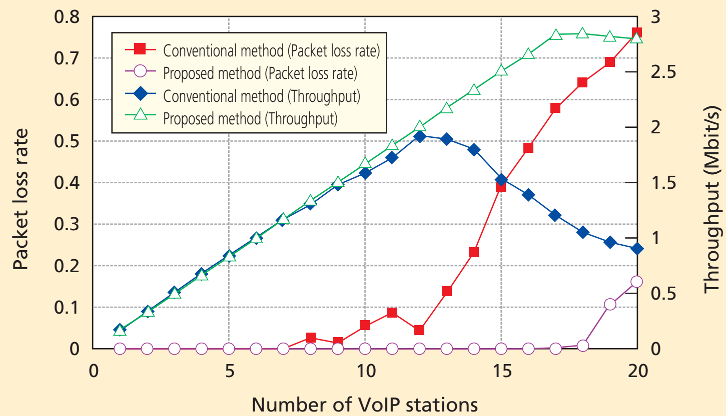

Figure 11 Simulation results for packet loss rate when number of VoIP stations increases

panies a higher number of simultaneous calls, which was an issue for the conventional method.

The proposed method can work well even when cells overlap in the same frequency band and packets of multiple voice codec periods are mixed together [13].

5. Conclusion

We have proposed an optimal AP selection method that takes QoS control into account and an autonomous distributed scheduling MAC method as technology for improving voice quality in wireless LAN VoIP. Future work includes testbed verification and preparation towards standardization.

REFERENCES

[1] IEEE 802.11e, 2005.

[2] S. Mangold, S. Choi, G. Hiertz, O. Klein and B. Walke: "Analysis of IEEE 802.11e for QoS Support in Wireless LANs," IEEE Wireless Communications, Vol. 10, No. 6, pp. 40–50, Dec. 2003.

[3] H. w. Lee, S. h. Kim and W. Ryu: "Performance of an efficient method for association admission control in public wireless LAN," VTC 2004 Fall.

[4] S. Misra and A. Banerjee: "A Novel Load Sensitive Algorithm for AP Selection in 4G Networks," CODEC 2003, Calcutta, INDIA, 2003.

[5] S. Vasudevan, K. Papagiannaki, C. Diot, J. Kurose and D. Towsley: "Facilitating access point selection in IEEE 802.11 wireless networks," in ACM Internet Measurement Conf., New Orleans, LA, Oct. 2005.

[6] W. M. Moh, D. Yao and K. Makki: "Wireless LAN: study of hidden station effect and multimedia support," Proc. Of 7th International Conference on Computer Communications and Networks, pp. 422–431, 12–15 Oct. 1998.

[7] L. Du, Y. Bai and L. Chen: "Access Point Selection Strategy for Large-Scale Wireless Local Area Networks," WCNC2007.

[8] IEEE P802.11v D0. 12, May. 2007.

[9] L. Du, Y. Bai, L. Chen, D. Gu, S. Choi, S. Jin, F. Watanabe, M. Jeong and A. Yamada: "QoS-aware Load Balancing," IEEE802.11–07/0326, Mar. 2007.

[10] X. Ling and K. L. Yeung: "Joint access point placement and channel assignment for 802.11 wireless LANs," in IEEE Wireless Communications and Networking Conference (WCNC), Mar. 2005.

[11] G. Hiem, S. Mangold and L. Berlemann: "QoS Support as Utility for Coexisting Wireless LANs," In Proceedings of the International Workshop on IP Based Cellular Networks, IPCN, Paris, France, Apr. 2002.

[12] K. Igarashi, A. Yamada, A. Fujiwara and T. Sugiyama: "Improving Wireless LAN VoIP Quality with Autonomous and Distributed Control between Stations," IEICE Technical Report, RCS2006–12, Apr. 2006.

[13] K. Igarashi, A. Yamada and T. Sugiyama: "Improving VoIP call quality in a multi-cell environment wireless LAN," IEICE Technical Report, RCS2006–11, Nov. 2006.