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VoIP Session Capacity Expansion with Packet Transmission Suppression Control in Wireless LAN

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SUMMARY   This paper proposes a VoIP (Voice over Internet Protocol) session capacity expansion method that uses periodic packet transmission suppression control for wireless LANs. The proposed method expands the VoIP session capacity of an AP without critically degrading the QoS (Quality of Service) of all stations. Simulation results show the proposed method with 0.5% packet suppression control on each station expands a VoIP session capacity by up to 5% compared to a legacy method while satisfying required QoS for all stations.
key words: *VoIP, wirelss LAN, capacity expansion, packet suppression*

1.   Introduction

Nowadays, WLANs (Wireless Local Area Networks), such as IEEE802.11a, 11g, 11n [1]–[3], have become popular. These WLAN technologies provide high-speed Internet accesses and enable us to use many and various multimedia services, such as VoIP (Voice over Internet Protocol), VoD (Video on Demand) or SaaS (Software as a Service), for example, on Internet via wireless access systems as well as conventional wired access systems. VoIP services have gotten a lot of attention recently in these multimedia services in order to provide inexpensive voice communications via Internet. However, VoIP services on Internet are less suitable than best-effort services, such as HTTP (Hyper Text Transfer Protocol) or FTP (File Transfer Protocol), for example, because there are no guarantees of QoS (Quality of Service) for VoIP services to transmit packets correctly in real-time. Especially in WLANs, it is more difficult to provide QoS for VoIP services because in wireless environments where is much interference from other wireless instruments, for example, and degradation factors, such as fading and shadowing.

In order to provide QoS supports in WLAN, IEEE802.11e [4] has been standardized in 2005, which classifies packets according to required QoS and prioritizes packets that require high-quality transmissions. IEEE802.11e realizes prioritized QoS guarantees and enables to use VoIP services in WLANs.

However, not only in legacy IEEE802.11 series WLANs but also in IEEE802.11e WLANs, the QoS of VoIP is more likely to degrade due to a lot of packet collisions

and long delay time. These degradations are caused by CW, which decides packet transmission timings in CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance) access in order to many STAs (Stations) can be associated with an AP at the same time. STA firstly sets a back-off time according to CW, and then counts down a back-off time when a channel is idle in order to decide a transmission timing of a packet. When many STAs are associated with an AP, a small CW value causes a lot of collisions and high PLR (Packet Loss Rate) because a smaller CW value is likely to set same back-off time among associated STAs. On the other hand, a large CW value causes long delay time because a larger CW value is likely to set longer back-off time among associated STAs. Thus a capacity of an AP, which is the maximum number of associated STAs that can be satisfied QoS such as low PLR and short delay time is limited by CW value.

To improve an upper bound of a capacity of an AP, many methods have been proposed. These methods focus on an achievement of 0% PLR with using an effective scheduling [5] or packet aggregations [6]–[9]. However in view of an R-value [10], which is one of the most famous quality evaluation criteria of voice communications, it is not absolutely necessary to achieve 0% PLR. If an R-value, R, is more than 80, a user satisfaction is estimated as "Satisfied" and a QoS of voice communication is assumed as guaranteed [11]. Therefore, some PLR are allowed which can achieve $R \geq 80$ at least in view of an R-value.

This paper proposes a VoIP session capacity expansion method with packet transmission suppression control. The proposed method suppresses packet transmissions periodically, and creates transmission opportunities for transmission-failed packets that caused by packet collisions or long delay time. The proposed packet suppression method improves QoS of poor quality STAs while degrades QoS of high quality STAs within the QoS allowable range, $R \geq 80$. Then the proposed method decreases a variance of R-values among STAs without critically degrading the average R-values and makes all STAs satisfy required QoS, while a legacy method causes a large R-value variance and critically degrades the QoS of some STAs. Due to avoiding a critical QoS degradation with suppressing packet transmissions, the proposed method expands a VoIP session capacity of an AP without critically degrading users' satisfactions.

The rest of paper is organized as follows. Section 2 evaluates an allowable PLR which satisfies $R \geq 80$ and decides a maximum packet suppression ratio (MPSR) which

is a maximum ratio of periodic packet suppression without degrading the QoS. Section 3 describes the proposed capacity expansion method with suppressing packet transmissions and Sect. 4 evaluates performances of the proposed method compared to a Legacy method. Finally, Sect. 5 describes a conclusion.

2. Maximum Packet Suppression Ratio

In this section, an allowable PLR is evaluated from the point of an R-value, and an MPSR is revealed which is defined as a maximum packet suppression ratio without critically degrading the QoS of each STA.

An R-value is calculated with the E-model which is defined in ITU-T recommendation G.107 [10]. Although the number of parameters of the E-model is 20, it is difficult to measure all parameters. So this paper considers 2 important parameters, PLR and one-way delay time, and also considers 2 codec parameters, equipment impairment factor and packet-loss robustness factor. Other parameters are set as default values defined in G.107.

Table 1 shows relations between R-values and user satisfactions which is defined in ITU-T recommendation G.109 [11]. This paper targets $R \geq 80$, which estimates user satisfaction as "Satisfied," as described in Table 1.

Firstly, relations between R-values and 2 important parameters, PLR and one-way delay time, are evaluated. And then relations between the 2 important parameters and the number of STAs are evaluated. After that, a maximum allowable PLR without critically degrading QoS is evaluated, and an MPSR is defined from these evaluations.

2.1 Relations of the R-Value to PLR and One-Way Delay Time

PCM codec which is defined in ITU-T G.711 [12] is assumed as a voice codec of VoIP, whose parameters are shown in Table 2.

Figure 1 shows relations of R-values to PLR and one-way delay time, which are calculated with the E-model [10]. The equipment impairment factor, I_e, and the packet-loss robustness factor, Bpl, which are used in the E-model calculation and which specify codec characteristics, is set as values defined in ITU-T G.113 recommendation [13], $I_e = 0$ and $Bpl = 4.8$. The one-way delay time, T, is from 0 [ms] to 300 [ms]. When $T = 0$ [ms], the R-value, R, achieves $R = 93.2$ at $PLR = 0\%$ and $R = 80.0$ at $PLR = 0.65\%$. Table 3 shows the maximum allowable PLR which can achieve $R = 80$ against each delay time, T. With increasing the delay time, the maximum allowable PLR moves in parallel to minus direction, and it degrades PLR to 0.33% at $T = 300$ [ms].

2.2 Relations of PLR and One-Way Delay Time to the Number of STAs

In this section, relations of PLR and one-way delay time to the number of STAs are evaluated with QualNet computer simulations [14]. We evaluate PLR and one-way delay time in wireless domain as shown in Fig. 2, because almost all degradations of QoS are occurred in wireless domain. IEEE802.11g [2] is assumed as WLAN interface, whose parameters are shown in Table 4. G.711 PCM codec is assumed as voice codec of VoIP whose parameters are shown in Table 2. Figure 2 shows the our assumed simulation model. Two different situations of RTS/CTS (Request to Send/Clear to Send) are evaluated, one is the transmission with RTS/CTS (w/RTS/CTS) mode and another is without RTS/CTS (w/o RTS/CTS) mode. All wireless stations are placed randomly in 100 m × 100 m square area for each trial according to an uniform probability distribution. The number of trials is 1000.

Figure 3 and Table 5 show the simulation results. Table 5(a) shows average PLR and average one-way delay time at $N = 22$ and $N = 23$ for w/RTS/CTS mode, and Table 5
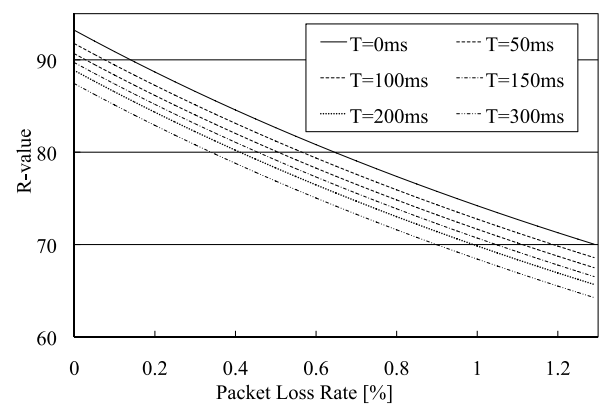
Table 1 The relations between the R-values and user satisfactions.

R-value (lower limit)	User satisfaction
90	Very satisfied
80	Satisfied
70	Some users dissatisfied
60	Many users dissatisfied
50	Nearly all users dissatisfied

Table 2 The parameters of G.711.

Transport layer Protocol	UDP
Payload Size	160 Byte
Interval	20 ms
Average Packet Arrival Rate	50 packets/sec
Required Data Rate	64 kbps
Direction	UP/DOWN

Fig. 1 The relations of the R-values to PLR and one-way delay time.

Table 3 The maximum allowable PLR for $R = 80$.

Delay time T[ms]	The maximum allowable PLR[%]
0	0.65
50	0.55
100	0.50
150	0.45
200	0.41
300	0.33
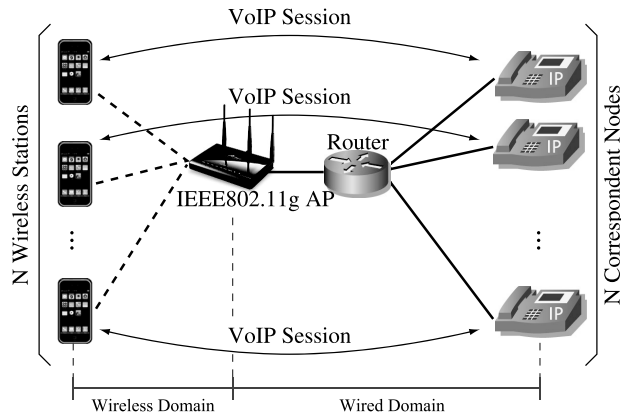
Fig. 2 Simulation model for evaluation the relation of PLR and delay time to the number of STAs.

Fig. 3 Average PLR and average one-way delay time with two-sided 99% confidence intervals to the number of STAs.

Table 4 IEEE802.11g simulation parameters.

Frequency	2.412 GHz
PLCP Preamble	16 μsec
PLCP Header (Signal)	1 Symbol
PLCP Header (Service)	16 bit
MAC Header	24 Octet
LLC Header	8 Octet
FCS	4 Octet
PLCP Tail	6 bit
Symbol Length	4 μsec
MAC ACK Length	10 Octet
Slot Time	9 μsec
SIFS	16 μsec
DIFS	34 μsec
CWmin	15
CWmax	1023
Short Retry Limit	7
Long Retry Limit	4

Table 5 Average PLR and average one-way delay time with two-sided 99% confidence intervals at (a) $N = 22$ and 23 for w/RTS/CTS, (b) $N = 27$ and 28 for w/o RTS/CTS.

(a) Transmisson with RTS/CTS		
	Average PLR	Average one-way delay
$N = 22$	$0.43 \pm 9.04 \times 10^{-5}$ %	112 ± 1.14 [ms]
$N = 23$	$6.47 \pm 1.27 \times 10^{-2}$ %	135 ± 1.35 [ms]

(b) Transmisson without RTS/CTS		
	Average PLR	Average one-way delay
$N = 27$	$0.24 \pm 1.00 \times 10^{-4}$ %	54 ± 1.26 [ms]
$N = 28$	$7.15 \pm 1.40 \times 10^{-2}$ %	113 ± 1.50 [ms]

(b) shows them at $N = 27$ and $N = 28$ for w/o RTS/CTS mode. Figure 3 and Table 5 also show two-sided 99% confidence interval for each result. These confidence intervals are calculated with t-distribution. With RTS/CTS mode results, the average PLR achieves 0.43% at $N = 22$, but increases drastically to 6.47% at $N = 23$. Collisions of up-link packets cause this high PLR. Many STAs with up-link flows send packets to an AP at the same time if back-off times are set as same values among many STAs due to small CW. The average one-way delay time increases to $T = 112$ [ms] at $N = 22$. Long back-off time and retransmission control cause this long delay time. Because a larger CW value is likely to set longer back-off time, many STAs with up-link flows wait for a long time to send packets. Retransmission control of transmission-failed packets also causes this long delay time. Also without RTS/CTS results mode, the average PLR achieves 0.24% at $N = 27$, but it increases to 7.15% at $N = 28$. The average one-way delay time 54 [ms] and 113 [ms] at $N = 27$ and $N = 28$, respectively. These degradations are also caused by up-link collisions and long back-off time.
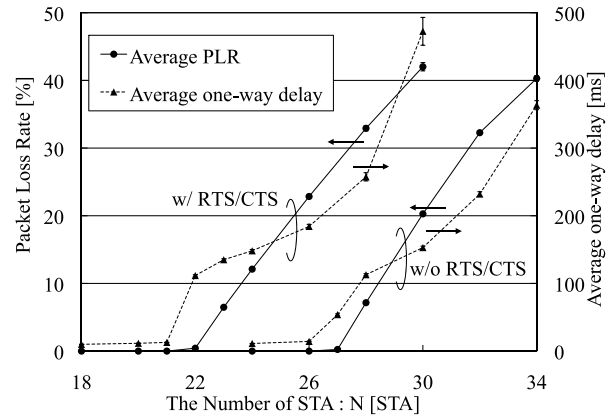
2.3 MSPR Evaluation

As shown in Fig. 3 and Table 5, PLR of VoIP packet transmissions with RTS/CTS are occurred in $N \geq 22$, especially in $N \geq 23$, and PLR without RTS/CTS are occurred in $N \geq 27$, especially in $N \geq 28$.

Firstly, we evaluate an MSPR for w/RTS/CTS mode. At $N = 22$, the average PLR is 0.43% and the average one-way delay time is 112 [ms]. From the parameters of G.711, as shown in Table 2, the number of transmission packets per second is 50, then the average number of transmission-failed packets is 0.215 per an STA per second, in other words, 4.73 packets per 22 STAs per second. Then we assume 0.50% suppression of packet transmissions, which can satisfies $R = 80$ at $T = 112$ [ms], calculated with the E-model. Because the number of transmission suppressed packets is 0.25 per an STA per second, in other words, 5.5 packets per 22 STAs per second, there is enough room for transmissions of all transmission-failed packets in legacy VoIP transmissions. Therefore a packet suppression method improves PLR and QoS for poor quality STAs while degrades QoS for good quality STAs within the QoS allowable range. Then it expands VoIP session capacity of an AP with satisfying the QoS of all STAs.

At $N = 23$, the average number of transmission-failed packets is 74.4 per 23 STAs per second and an average one-way delay time is 135 [ms]. Then we assume 0.45% packet

suppression, which can satisfy $R = 80$ at $T = 135$ [ms], calculated with the E-model. Although the number of transmission suppressed packets is 5.4 per 24 STAs per second, there is not enough room for transmissions of all transmission-failed packets in legacy VoIP transmissions. So if there are over 22 STAs, a packet transmission suppression control is not effective without critically degrading the QoS.

Next, we evaluate an MSPR for w/o RTS/CTS mode. At $N = 27$, due to 0.24% average PLR, the average number of transmission-failed packets is 3.24 per 27 STAs per second. Then we assume 0.50% suppression of packet transmissions, which enough satisfies $R \geq 80$ at an average one-way delay time, $T = 54$ [ms]. Because the number of transmission suppressed packets is 6.75 per 27 STAs per second, there is enough room for transmissions of all transmission-failed packets in legacy VoIP transmissions. Therefore a packet suppression method expands a VoIP session capacity of an AP without critically degrading the QoS.

At $N = 28$, the average number of transmission-failed packets is 100.1 per 28 STAs per second and the average one-way delay time is 113 [ms]. Then we assume 0.50% packet transmission suppression, which satisfies $R = 80$ at $T = 113$ [ms], calculated with the E-model. Although the number of transmission suppressed packets is 7 per 28 STAs per second, there is not enough room for transmissions of all transmission-failed packets. So if there are over 28 STAs, a packet transmission suppression control is not effective without critially degrading the QoS.

Based on the above evaluations, we target to show effectiveness of a packet transmission suppression control at $N = 22$ for w/RTS/CTS mode and at $N = 27$ for w/o RTS/CTS mode. For w/RTS/CTS mode, maximum allowable PLR is 0.5% which are from the average one-way delay time is 112 [ms] at $N = 22$. On the other hand, for w/o RTS/CTS mode, maximum allowable PLR is also 0.5% which enough satisfies $R \geq 80$ at the average one-way delay time is 54 [ms] of $N = 27$. Therefore, we define MSPR as 0.5%, which is enough effective in both RTS/CTS modes.

3. VoIP Session Capacity Expansion Method with Packet Transmission Suppression Control

In this section, we describe the proposed VoIP session capacity expansion method with packet transmission suppression control. This proposed method suppresses packet transmissions periodically for each STA in order to prevent a capacity of an AP from reaching to critical limit. This method creates transmission opportunities for transmission-failed packets due to collisions or long delay time and it expands the VoIP capacity without critically degrading the QoS of all STAs.

In this proposed method, each STA feeds back the number of its own retransmission time of a packet to around STAs with appending information to packets. Each STA suppresses packet transmissions within the QoS allowable range that depends on the fed back information. Due to this packet transmission suppression control, the proposed
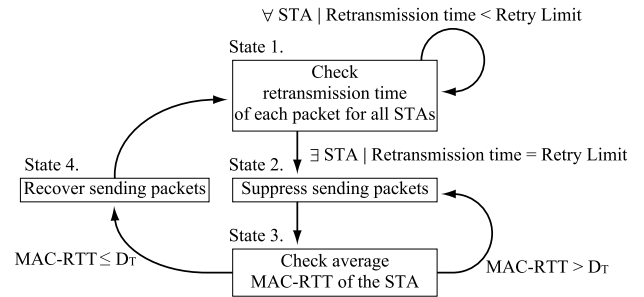

Fig. 4 Proposed packet transmission suppression control algorithm.

method makes packet transmission rooms for poor quality STAs with suffering from high PLR and long delay time, reduces the gap of QoS among STAs and expands the VoIP capacity without critically degrading the QoS of all STAs. Fig. 4 shows the proposed packet transmission suppression control algorithm, as follows.

State 1

An STA gets information of the number of retransmission time via channel scanning. Each STA appends the information of the number of retransmission time to each sending packet. If an STA uses RTS/CTS mechanism, the retransmission information is appended to each RTS and CTS packet, otherwise it is appended to each data packet. The appended information size is generally 1 octet because the maximum retransmission time is 7 in general IEEE802.11 series WLANs. After an STA gets the retransmission time information from all around STAs, if the number of retransmission time of all STAs are less than Retry Limit of IEEE802.11, which is defined as the upper bound of the number of retransmission time, then an STA continues to scan the channel and returns to State 1. On the other hand, if there exists an STA whose retransmission time reaches Retry Limit, an STA estimates congestions. Then an STA moves to State 2 in order to avoid congestions and expand a capacity of an AP.

State 2

An STA suppresses packet transmissions periodically by a ratio of α%, then moves to State 3.

State 3

An STA observes its own MAC-RTT (Media Access Control layer Round Trip Time) of each sending packet. MAC-RTT is defined as a duration from an STA beginning of back-off countdown to the STA receiving MAC ACK (Acknowledgement) frame, as shown in Fig. 5. If an average MAC-RTT is more than a threshold D_T, an STA estimates there are still congestions and returns to State 2. If an average MAC-RTT is less than D_T, an STA estimates that congestions are avoided and moves to State 4.

State 4

An STA aborts a packet transmission suppression control and moves to State 1.
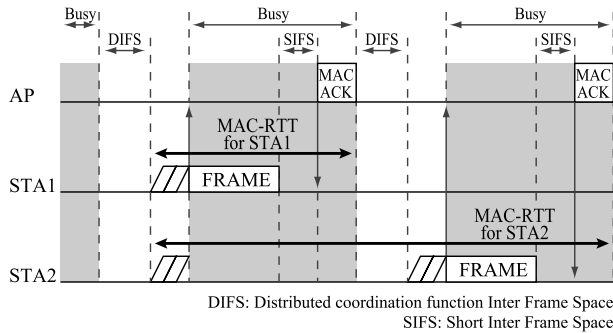
Fig. 5   MAC-RTT definition.

Fig. 6   Simulation model for performance evaluations.

4.   Performance Evaluation

The proposed method (Proposed) performances are evaluated compared to a legacy method (Legacy) which is VoIP transmission via IEEE802.11g without suppressing packet transmissions.

4.1   Simulation Model

A simulation model is shown in Fig. 6. All STAs are placed randomly according to an uniform probability distribution. We assume IEEE802.11g as WLAN interface whose parameters are shown in Table 4 in Sect. 2.2. G.711 PCM codec is assumed as VoIP codec, whose parameters are shown in Table 2 also in Sect. 2.2.

We evaluate two different modes, w/RTS/CTS mode and w/o RTS/CTS mode, which is same description as Sect. 2.2. If an STA uses RTS/CTS, an STA can avoid a hidden stations problem. But using RTS/CTS with smaller size packets is less effective than with larger size packets. Contrary, if an STA doesn't use RTS/CTS, an STA can achieve more effective frequency utilization due to smaller overheads.

A packet suppression ratio, α, is 0.5%, which is the MSPR evaluated in Sect. 2.3. An MAC-RTT threshold, D_T, is 200 [ms] and 100 [ms], which are twice time of one-way average delay time of improvement targets for w/RTS/CTS mode and w/o RTS/CTS mode, respectively. We assume an appended information size of retransmission time as 1 octet, which can be enough to indicate Long Retry Limit that is maximum retransmission time of RTS/CTS mode in this simulation model.

We evaluate an average PLR, an average one-way delay time, an average R-value and a VoIP session capacity of an AP. A VoIP session capacity of an AP is evaluated by a QoS achievement STA ratio, which is a ratio of STAs with satisfying the required quality, $R \geq 80$. We defines a VoIP session capacity of an AP as a maximum number of STAs that satisfies a QoS achievement ratio is more than 95% [6].

We also evaluate two-sided 99% confidence interval of each evaluation with t-distribution. The number of trials is 1000.
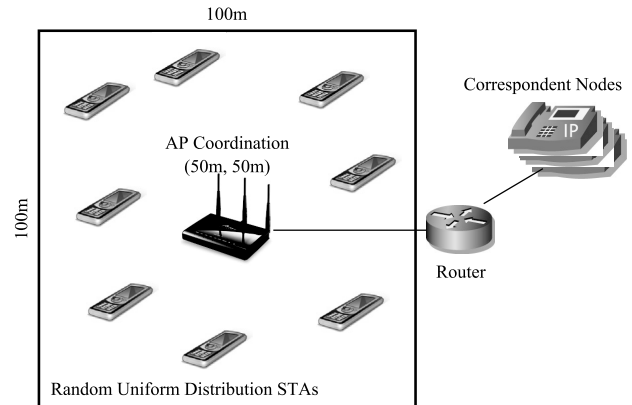

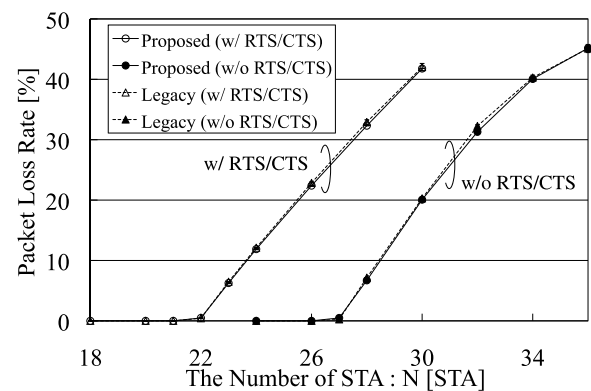**Fig. 7**   Average PLR to the number of STAs.

Table 6   Average PLR with two-sided 99% confidence intervals at (a) $N = 22$ and 23 for w/RTS/CTS, (b) $N = 27$ and 28 for w/o RTS/CTS.

(a) Transmisson with RTS/CTS

	Proposed	Legacy
$N = 22$	$0.481 \pm 6.96 \times 10^{-20}\%$	$0.427 \pm 9.04 \times 10^{-5}\%$
$N = 23$	$6.26 \pm 2.76 \times 10^{-7}\%$	$6.47 \pm 1.27 \times 10^{-2}\%$

(b) Transmisson without RTS/CTS

	Proposed	Legacy
$N = 27$	$0.481 \pm 7.65 \times 10^{-20}\%$	$0.241 \pm 1.00 \times 10^{-4}\%$
$N = 28$	$6.71 \pm 3.00 \times 10^{-7}\%$	$7.15 \pm 1.40 \times 10^{-2}\%$

4.2   Average PLR Evaluation

Figure 7 shows average PLR and confidence intervals to the number of STAs and Table 6 shows average PLR with two-sided 99% confidence intervals at $N = 22$ and 23 for w/RTS/CTS, and $N = 27$ and 28 for w/o RTS/CTS. From the average PLR point of view, there are almost no differences between Proposed and Legacy in both RTS/CTS modes.

At the range of $N \leq 21$ in w/RTS/CTS mode and $N \leq 26$ in w/o RTS/CTS mode, due to not suppressing packet transmissions, Proposed and Legacy achieve same average PLR, 0%.
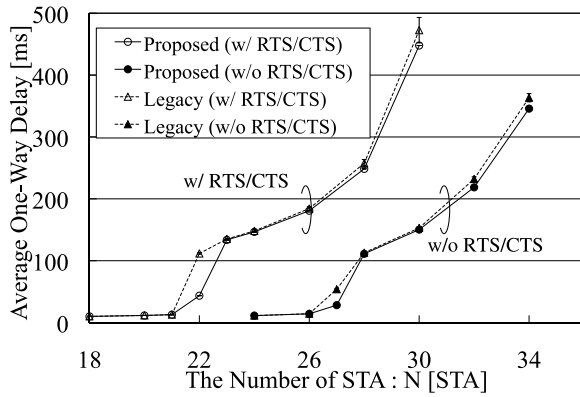
Fig. 8   Average one-way delay time to the number of STAs.

4.3   Average One-Way Delay Time Evaluation

Figure 8 shows an average one-way delay time and confidence intervals to the number of STAs and Table 7 shows average one-way delay time with two-sided 99% confidence intervals at $N = 22$ and 23 for w/RTS/CTS, and $N = 27$ and 28 for w/o RTS/CTS. At $N = 22$ in w/RTS/CTS mode and $N = 27$ in w/o RTS/CTS mode, Proposed shortens the average one-way delay time by 61% and 48% compared to Legacy, respectively. Proposed transmits less packets than Legacy at $N = 22$ or $N = 27$, then it makes a room for packet transmissions to reduce the number of retransmission time. Because wait time of retransmission is a major factor of the average one-way delay time in WLAN, Proposed shortens the average one-way delay time.

On the other hand, at the range of $N \geq 24$ and $N \geq 28$ in w/RTS/CTS mode and w/o RTS/CTS mode, respectively, Proposed shorten the delay time by about 10% compared to Legacy. This is because the same reason of above case. Proposed can reduce the number of retransmission and shorten the average one-way delay time.

Except for the above cases, Proposed and Legacy achieve almost same average one-way delay time. At the range of $N \leq 21$ and $N \leq 26$ in w/RTS/CTS mode and w/o RTS/CTS mode, respectively, due to not suppressing packet transmissions, Proposed behaves the same as Legacy and achieves same average one-way delay time as Legacy.

4.4   Average R-Value Evaluation

Figure 9 shows an average R-value and confidence intervals

Table 7   Average one-way delay time with two-sided 99% confidence intervals at (a) $N = 22$ and 23 for w/RTS/CTS, (b) $N = 27$ and 28 for w/o RTS/CTS.

(a) Transmisson with RTS/CTS

	Proposed	Legacy
$N = 22$	$43.5 \pm 6.22 \times 10^{-2}$ ms	111.7 ± 1.14 ms
$N = 23$	$133.7 \pm 1.38 \times 10^{-2}$ ms	135.0 ± 1.35 ms

(b) Transmisson without RTS/CTS

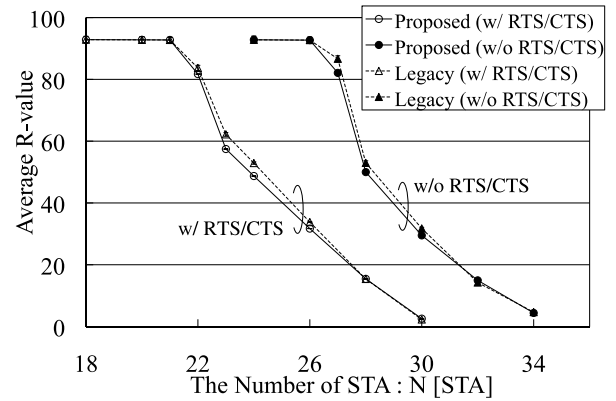|	Proposed	Legacy
$N = 27$	$28.2 \pm 6.84 \times 10^{-2}$ ms	53.9 ± 1.26 ms
$N = 28$	$111.2 \pm 1.50 \times 10^{-2}$ ms	112.3 ± 1.50 ms

Fig. 9   Average R-value to the number of STAs.

Table 8   Average R-value with two-sided 99% confidence intervals at (a) $N = 22$ and 23 for w/RTS/CTS, (b) $N = 27$ and 28 for w/o RTS/CTS.

(a) Transmisson with RTS/CTS

	Proposed	Legacy
$N = 22$	$81.7 \pm 4.92 \times 10^{-5}$	$83.7 \pm 8.72 \times 10^{-1}$
$N = 23$	$57.5 \pm 4.52 \times 10^{-5}$	$62.2 \pm 5.90 \times 10^{-1}$

(b) Transmisson without RTS/CTS

	Proposed	Legacy
$N = 27$	$82.1 \pm 5.40 \times 10^{-5}$	86.7 ± 1.00
$N = 28$	$50.0 \pm 5.00 \times 10^{-5}$	$53.0 \pm 6.50 \times 10^{-1}$

to the number of STAs and Table 8 shows average R-value with two-sided 99% confidence intervals at $N = 22$ and 23 for w/RTS/CTS, and $N = 27$ and 28 for w/o RTS/CTS. At the range of $N \leq 21$ and $N \leq 26$ of w/RTS/CTS mode and w/o RTS/CTS mode, respectively, Proposed and Legacy achieve the same R-values. Due to not suppressing packet transmissions, Proposed behaves the same as Legacy.

On the other hand, at the range of $N \geq 22$ and $N \geq 27$ of w/RTS/CTS mode and w/o RTS/CTS mode, respectively, Proposed degrades average R-values up to 8% compared to Legacy. Due to packet transmission suppression, Proposed degrades R-values compared to Legacy whose degradation is mainly affected by PLR. In spite of average R-value degradations, Proposed improves a VoIP session capacity compared to Legacy. This capacity expansion issue
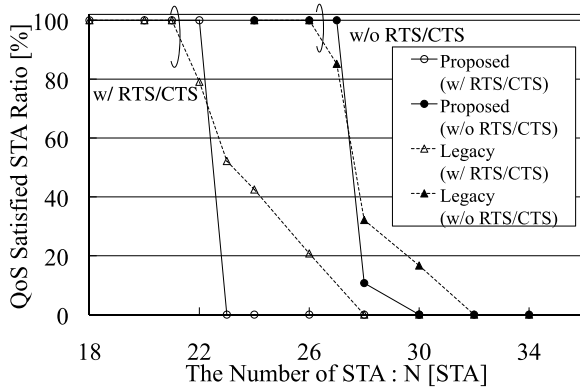
Fig. 10   QoS satisfied STA ratio to the number of STAs.

Table 9   QoS satisfied STA ratio with two-sided 99% confidence intervals at (a) $N = 22$ and 23 for w/RTS/CTS, (b) $N = 27$ and 28 for w/o RTS/CTS.

(a) Transmisson with RTS/CTS

	Proposed	Legacy
$N = 22$	100.0 ±0%	79.1 ±4.97 × 10^{-5}%
$N = 23$	0.00 ±0%	52.2 ±1.32 × 10^{-5}%

(b) Transmisson without RTS/CTS

	Proposed	Legacy
$N = 27$	100.0 ±0%	85.2 ±5.21 × 10^{-1}%
$N = 28$	10.71 ±0%	53.0 ±4.25 × 10^{-1}%

Table 10   VoIP session capacity of an AP for each method.

Method	VoIP session capacity
Proposed (w/RTS/CTS)	22
Legacy (w/RTS/CTS)	21
Proposed (w/o RTS/CTS)	27
Legacy (w/o RTS/CTS)	26

is discussed in next section.

4.5   VoIP Session Capacity Evaluation

Figure 10 shows a QoS satisfied STA ratio and confidence intervals to the number of STAs and Table 9 shows QoS satisfied STA ratio with two-sided 99% confidence intervals at $N = 22$ and 23 for w/RTS/CTS, and $N = 27$ and 28 for w/o RTS/CTS. Table 10 shows a VoIP session capacity of an AP of each method. In the case of w/RTS/CTS mode, Proposed keeps a QoS satisfied STA ratio more than 95% to 22 STAs, while Legacy keeps more than 95% to 21 STAs. On the other hand, also in the case of w/o RTS/CTS mode, Proposed keeps a QoS satisfied STA ratio more than 95% to 27 STAs, while Legacy keeps more than 95% to 26 STAs. Thus, Proposed improves a VoIP session capacity of an AP, which is an upper bound of the number of STAs with satisfying more than 95% of QoS satisfied STA ratio, as shown in Table 10, by 5% and 4% for w/RTS/CTS mode and w/o RTS/CTS mode, respectively.

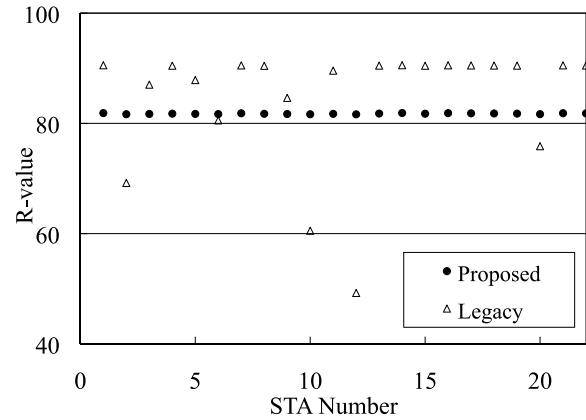This is because that Proposed achieves a fairness R-

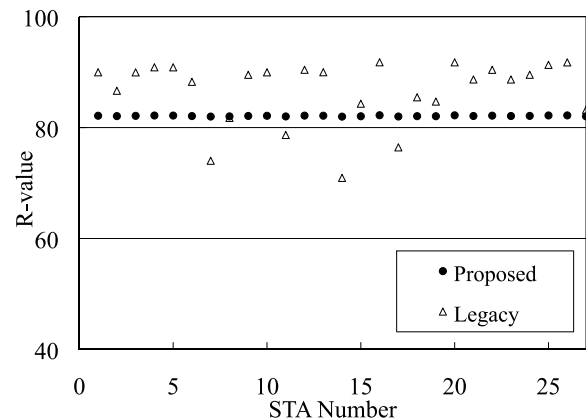**Fig. 11**   R-value distribution at $N = 22$ of w/RTS/CTS.

Fig. 12   R-value distribution at $N = 27$ of w/o RTS/CTS.

value among STAs in each case. Figures 11 and 12 show the R-values distribution of STAs in each case of w/RTS/CTS mode and w/o RTS/CTS mode, respectively. In the case of w/RTS/CTS mode, as shown in Fig. 11, Proposed achieves R-values of around $R = 80$ for all STAs, while Legacy achieves R-values of around $R = 90$ for most STAs but achieves R-values of $R < 80$ for some STAs, especially achieves a lowest R-value of $R = 49.2$. This large R-value variance of Legacy causes a degradation of QoS satisfied STA ratio at $N = 22$. On the other hand, the small R-value variance of Proposed makes all STAs keep R-value more than 80. Also in the case of w/o RTS/CTS mode, as shown in Fig. 12, Proposed achieves R-values around $R = 80$ for all STAs, while Legacy achieves R-values around $R = 90$ for most STAs but around $R = 70$ for some STAs. So Legacy degrades a QoS satisfied STA ratio.

Due to suppressing packet transmissions without critically degrading the QoS of all STAs, Proposed improves an average PLR and shortens an average delay time for lower QoS STAs, while degrades them for higher QoS STAs within the QoS allowable range. So Proposed prevents critical R-value degradations for some STAs, improves a QoS satisfied STA ratio and then expands a VoIP session capacity of AP by 5% and 4% for w/RTS/CTS mode and w/o

Table 11　Assumed average packet arrival rates.

	Packet Arrival Rate	Payload Size	Data Rate
80 PPS	80 packets/sec	160 Bytes	102.4 kbps
50 PPS	50 packets/sec	160 Bytes	64 kbps
40 PPS	40 packets/sec	160 Bytes	51.2 kbps
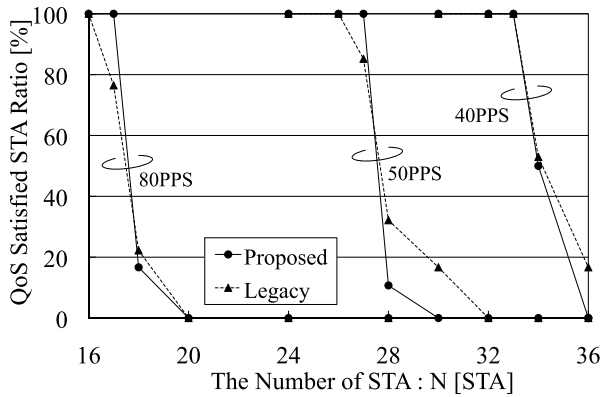
Fig. 13　QoS satisfied STA ratio to the number of STA with different packet arrival rates.

Fig. 14　QoS satisfied STA ratio of Proposed-Hybrid is indicated by black line, that of Proposed and Legacy are indicated by gray line.

RTS/CTS mode, respectively.

We also evaluated QoS satisfied STA ratio for w/o RTS/CTS mode with different average packet arrival rates. 3 different average packet arrival rates are assumed as shown in Table 11. Figure 13 shows QoS satisfied STA ratios of 3 different average packet arrival rates, 80, 50 and 40 PPS. If an average packet arrival rate is high, such as 80 PPS, Proposed is more effective than Legacy. But if an average packet arrival rate is low, such as 40 PPS, Proposed is as effective as Legacy, but not less effective than Legacy. So, Proposed is more effective with higher average packet arrival rate. This result is also same as w/RTS/CTS mode.
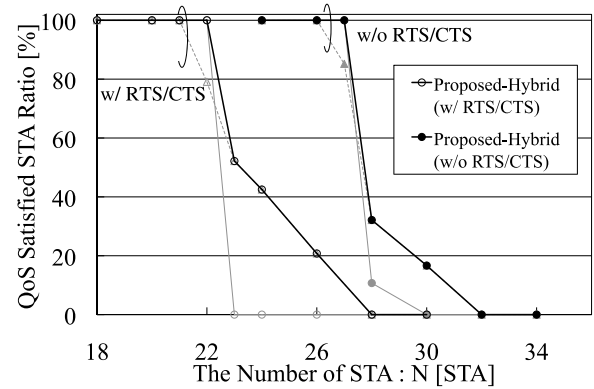
4.6　Hybrid Packet Suppression Method

As shown in Fig. 10, Proposed decreases the QoS satisfied STA ratio more drastically than Legacy. Proposed loses R-value more easily than Legacy when some packet losses are occurred because Proposed suppresses transmission packets up to the QoS allowable limit. So if a user requires at least some of STAs satisfying QoS although most of STAs can't satisfy it, Proposed is less effective than Legacy. If there is such requirement, Proposed needs to add a state transition condition to State 3 in the algorithm that is described in Sect. 3, as follows.

State 3

An STA observes its own packet loss. If an STA detects a packet loss due to congestions, an STA estimates that only a packet suppression control can't avoid congestions and that it makes a lot of packet losses and degrades QoS. Then an STA moves to State 4 in order to reduce packet loss.

This method is Proposed and Legacy hybrid method (Proposed-Hybrid).　It suppresses packet transmissions

when Legacy degrades QoS satisfied STA ratio, on the other hand, when Proposed degrades QoS satisfied STA ratio, it aborts suppressing packet transmissions. Since Proposed-Hybrid incorporates good parts of Proposed and Legacy, it is more effective than Proposed in point of the requirement that at least some of STAs satisfying QoS although most of STAs can't satisfy it. Figure 14 shows QoS satisfied STA ratio of Proposed-Hybrid, which is obtained from theoretical calculation that selects a larger value of QoS satisfied STA ratio between Proposed and Legacy. As shown in Fig. 14, Proposed-Hybrid improves not only VoIP session capacity compared to Legacy but also QoS satisfied STA ratio above the capacity limit of an AP compared to Proposed.

5.　Conclusion

A VoIP session capacity expansion method with packet transmission suppression control is proposed. This proposed method suppresses 0.5% packet transmission periodically, then the proposed method improves QoS of poor quality STAs while degrades QoS for high quality STAs within the QoS allowable range. The simulation results show the proposed method expands a VoIP session capacity of an AP by up to 5% compared to a legacy method. And the proposed method expands more capacity when an average packet arrival rate is more higher.

A hybrid packet suppression method is also proposed. This hybrid method incorporates good parts of the proposed method and the legacy method. A theoretical calculation shows the hybrid method improves not only a VoIP session capacity but also a QoS satisfied STA ratio.

This proposed method can uses in combination with these methods because this proposed method is independent of other capacity expansion methods such as packet aggregation methods or effective packets scheduling methods. If the proposed method uses in combination with these methods, more capacity expansions can be expected. This combination methods evaluation is further study.

Acknowledgements

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