YAMAHA

PJP-50R

F.W. 2.10

Additional information

The configuration saving and loading features

Due to changes in specifications, the configuration saving and loading features have been newly added. These features are useful when you want to use this unit temporarily in other locations. You can save up to 3 configurations.

■ Saving settings

Using the keys on this unit

1 In the initial display, select "Menu"→ "Settings" → "Save/Load Config" → "Save Config".

2 From the three configuration areas, select the one you want to save settings in.

Using the Web menu

1 Select "Save Config".

2 From the three configuration areas, select the one you want to save settings in.

Notes

- If you select a configuration that has already been saved, the saved configuration will be overwritten.
- The address book, the history of calls, and the configured password are not saved.
- When you restoring the factory setting by using "Resetting this Unit" (page 42 in Setup Guide), this unit also deletes the saved settings.

■ Loading settings

Using the keys on this unit

1 In the initial display, select "Menu"→ "Settings" → "Save/Load Config" → "Load Config".

2 From the three configuration areas, select the one you want to load settings from.
The current configuration is overwritten by the selected configuration, and this unit restarts.

Using the Web menu

1 In the Web menu, select "Load Config".

2 From the three configuration areas, select the one you want to load settings from.
The current configuration is overwritten by the selected configuration, and this unit restarts.

SIP address in the initial display (Setup Guide: page 24)

With the change in the specifications, you can check the SIP address set for the ProjectPhone in the initial display. Press to display the IP and SIP address. The addresses appear in a few seconds.

Sending tone signals during communication through IP network (Basic Operation Guide: page 13)

Tone signals can be transmitted during communication through IP network. This is convenient when using a service that requires tone input during a call, such as the automatic answering service.

1 Press during communication through IP network.
The display for additional number entry appears.

2 Press #, ∗, or the numeric keys to enter the required number.
The corresponding tone is transmitted.

If this unit is in multiple connections, select the connection you want to send tone signals in the "Select NumberAddition" display, and press .

3 After completing the number entry, press .
The display for communication reappears.

Note

While sending tone signals, this unit cannot receive any incoming calls.

QoS compatibility

With the change in the specifications, a QoS (Quality of Service) setting has been added. It is possible to specify the ToS (Type of Service) or CoS (Class of Service) priority. When CoS is used, the VLAN-ID can also be set.

Using the Web menu

In the Web menu, Select "QoS".

G.729a codec compatibility (Setup Guide: page 19)

G.729a codec is available because of the change of the specification.

Using the keys on this unit

In the initial display, select “Menu” → “Settings” → “Sound Settings” → “CODEC” → “G.729”.

Using the Web menu

In the Web menu, select “CODEC” and then “G.729”.

RTP packet transmission interval setting (Setup Guide: page 19)

With the change in the specifications, a function for setting the RTP (Real-time Transport Protocol) packet transmission interval has been added.

- **G.711 Extension**: 20 msec (Default), 40 msec
- **G.711**: 20 msec (Default), 40 msec
- **G.726-32**: 20 msec (Default), 40 msec
- **G.729**: 10 msec, 20 msec, 40 msec (Default), 60 msec

Using the Web menu

In the Web menu, select “CODEC” and then set the encoding method and “RTP Segment Size”.

About settable SIP addresses (Setup Guide: page 24)

With the change in the specifications, it is now possible to set SIP addresses including the # and ∗ symbols. The number of characters that can be set for the SIP address has also been changed to up to 32 characters each for the user ID section and domain section. If an SIP address including more than 32 characters in the user ID section or domain section is set, the restriction to the number of characters is applied the next time the registration is modified.

SIP registration expiration / session expiration setting (Setup Guide: page 24)

With the change in the specifications, it is now possible to set SIP registration expiration and session expiration.

Using the Web menu

In the Web menu, select “SIP Server” and then set “Regist Expiration” and “Session Expiration”.

Setting the country using the telephone function (Setup Guide: page 16)

With the change in the specifications, the name of the menu for setting the country using this unit has been changed from “Country Code” to “Area Code”. In addition, the following area has been added.

Australia, HongKong,Macao, Korea, Taiwan

Setting the telephone circuit mode (Setup Guide: page 16)

With the change in the specifications, the content of this setting has been changed from selecting the telephone circuit mode when using this unit in Japan or China to Japan or the other countries. Due to this change, “Mode 3” has been added.

- **Mode 3**: Use this mode if the extension telephones do not respond properly to an incoming call in “Mode 1”and “Mode 2”. The incoming call ringer rings without checking the CND (Caller Number Display) or CID (Caller ID) data from your telephone company.

When “Setting the country using the telephone function” is set to other than “Japan”, changing this setting between “Mode 1” and “Mode 2” does not affect the function of this unit.

On-hook dial feature

With the change in the specifications, it is now possible to enter the address after pressing key. This is convenient when calling the other locations during a call.

1 While the initial display or calling display is being displayed, press .

“Number” screen appears, and then a dial tone is output.

2 Enter the address with the numeric keys.

The destination telephone starts to be called.

Using the keys on this unit

In the initial display, select “Menu” → “Settings” → “General Settings” → “Manual Call” → “Enable”

Using the Web menu

In the Web menu, select “General Settings” → “Manual Call” → “Enable”

- In “Dialing Timeout” in the Web menu, it is possible to set the time from when finishing to enter the address till when this unit starts calling (1-10 seconds). The default setting is 3 seconds.
- In the initial display, it is also possible to start calling by pressing key after entering the address.

Suppressing the telephone echo

With the change in the specifications, it is now possible to activate or deactivate the telephone echo suppressor when starting a call.

- **Enable** (default): Select to activate the telephone echo suppressor.
- **Disable**: Select to deactivate the telephone echo suppressor.

Using the keys on this unit

In the initial display, select “Menu” → “Settings” → “Phone” → “ Echo Suppressor”.

Using the Web menu

In the Web menu, select “ Telephone Settings” → “ Echo Suppressor”

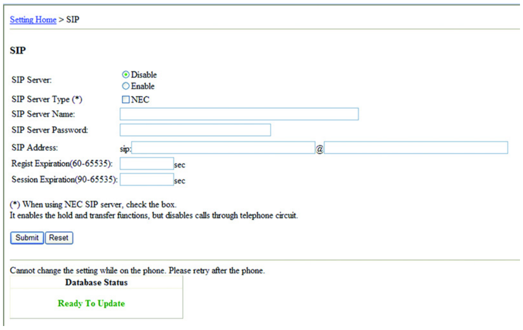
Using the NEC Mode

With the change in the specifications, NEC Mode has been added. When using this unit with NEC SV7000 or SV8500, it is now possible to hold and transfer a call during it.

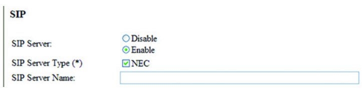
■ Setting the NEC Mode

Using the Web menu

1 In the Web menu, select “SIP”.



2 In “SIP Server”, select “Enable”, and then check “NEC” check box in “SIP Server Type”.



■ Using the hold/transfer function

1 Press to hold a call during it.

“Holding” screen appears. Press again to resume the call that has been held.

2 Select a transfer destination from “Call History” or “Online”, and then press , or enter telephone number of a transfer destination with the numeric keys, and then press .

The transfer destination telephone starts to be called.

Notes

- Communications through the telephone network and talks with multiple locations are not available while using the hold/transfer function.
- The audio is output in monaural despite the setting of “Speaker Mode”.

- When entering telephone number with the numeric keys, the transfer destination telephone also starts to be called after a certain period of time. In “Dialing Timeout” in the Web menu, it is possible to set the time from when finishing to enter the address till when this unit starts calling (1-10 seconds). The default setting is 3 seconds.
- While calling the transfer destination, the transfer turns to the quick transfer mode (transfers before the transfer destination answers the call) by pressing . The call is canceled by pressing key.