Release Notes
7800VDPX & 7800VDOX
2.22f

Enhancements

- Support for PSTN Answering machine (recording time is 60sec)
  *#90 : Access the PSTN A/M
  Note: 78V uses the 1st available phone port to record the PSTN message. So, the A/M will stop re-cording if user picks up the specified phone.
- Support for Phone Book/Speed Dial. The speed dial code is **1xx where xx can be set in the Phone Book.
- Remove the timer (20sec) to terminate VoIP unanswered call
- Add timer settings (0, 5, 10, 15, 20 sec) for A/M to answer the call
- Support for VoIP Fax Reception and Fax to email
- Support for GRE VPN (7800VDOX only)
- Hide VoIP password
- Support for 1-1 NAT
- If the device IP is obtained from DHCP, other devices can access the device through via device name.
- Add "Host Label" field on the Fixed Host page and WOL page. Rename "User Name" to "Host Label" at the Time Restriction page.
- Add PPTP default gateway (default route) selection to the PPTP client section(7800VDOX only).
- Allow X wildcard character with PTSN dial plan.
- Maximum number of Virtual Server rules allowed increased from 32 to 64.

Fixed

- Fixed the IP range issue with the DHCP server (for example, if the router IP is 192.168.1.253, 192.168.1.254-192.168.1.254 was previously only allowed for the range)
- Issue whereby dial plan might not work (buffer overflow) if there are more than 2 accounts under the same provider.
- Error with phone driver whereby phone may behave strangely if user picks up phone and then puts down without connection being made.
- Fixed issue whereby PPTP status page might be incorrect if more than one tunnel are manually discon-nected at the same time. (7800VDOX)
- MAC settings on relative pages should case be ignored.
- Fixed issue whereby PPTP client process doesn’t execute “connect” on the PPTP status page if the PPTP server fails to connect at the beginning. (7800VDOX)
- Fixed issue whereby MTU value on WAN Service page is not applied to the new one if user only edits the MTU value.

- Fixed issue whereby VoIP/PSTN sometimes doesn't work anymore whilst user dials a PSTN call and hook FLASHES to abandon the dialing. (VoIP process crasheing).
- Fixed issue with mail daemon behavior (add log to show the error case, shorten the connection fail timer, more robust the memory control to prevent from overflow) to prevent email task from hanging.
- Fixed issue whereby new VoIP settings can't work if user changes VoIP settings before the VoIP task finishes mail sending.
- Fixed issue whereby dnsproxy process might be gone and fail to dns query if the secondary wan profile is disconnected and main wan profile is still connected.