"FAQs on VFX88L"

1. Which signaling protocol supported by the SETU VFX?

Ans. SETU VFX supports SIP signaling protocol.

2. Which parameters we need to program to make calls from FXS port 2001 to FXS port 2002? We need to program following parameters in Jeeves to make calls from FXS port 2001 to FXS port 2002.

Ans.

- Enable SIP status in SIP account parameter1
- Set routing type called number based in SIP account parameter 2
- Program called number based routing table
- Program FXS port routing group

3. How to default SETU VFX?

Ans. You can default the SETU VFX through Jeeves. There is no hardware default available.

4. Is it possible to route call from SIP account to FXO port in SETU VFX?
Ans. No it's not implemented in this device.

5. Is it possible to connect Extension of PBX to FXO Port of SETU VFX for using PBX's trunk line from VoIP call?

Ans. No it's not implemented in this device.

6. How to check IP Address of SETU VFX?

Ans. We need to connect a CLI phone on FXS port to check an IP address.

- Enter into programming mode by #19 1234 (where 1234 is default password)
- Enter command 31#* (You will get conformation tone)
- Now ON Hook the phone
- You will get IP address on display with ring
- Now Off Hook the phone & dial 00#* to exit from programming mode

7. What is peer-peer calling?

Ans. Peer-peer calling is the same as IP-to-IP calling.
IP-to-IP call is possible in the following applications:

- Calls inside the same network from one LAN port to another LAN port
- Call inside the networks WAN, VLAN, VPN, RF and Satellite from one IP address to another IP address
- Calls between various networks

8. Why call is not getting through, when SETU VFX is configured in peer to peer communication?

Ans. This problem may arise if we have not enabled SIP status. So, enabling SIP status in SIP account parameters 1 menu of Jeeves may solve this issue.

9. We have an internet sharing with DHCP IP assignment & we want to use SETU VFX for peer to peer calling, but call is not getting through. So what is the problem?

Ans. We need to assign fix IP for peer to peer calling. So keep certain IP pool fix in DHCP server, which you can assign to SETU VFX.

10. I have Broadband connection. I want to use SETU VFX for peer to peer calling as well Internet. To get Internet on PC connected in LAN, I kept SETU VFX on Private IP & port forwarded from ADSL modem. Now problem is that, when I call from SETU VFX, call is established but no voice. How to solve this problem?

Ans.

- This problem may happen, if you are unable to receive speech packets
- STUN server gives information to the SETU VFX about which port is assigned by router to send speech packets
- You need to set source port IP address as IP address fetched using STUN & add STUN server address in Network parameter to solve this problem

11. What is the minimum bandwidth required for SETU VFX?

- The bandwidth required by VoIP ATA depends on the type of Vocoder used in the VoIP Box
- For example, the minimum bandwidth required is 21 Kbps if the Vocoder is G.723
- SETU VFX supports 6 types of Vocoders:
    1. G.711 (PCM) A-Law
    2. G.711 (PCM) µ-Law
    3. G.729
    4. G.723
    5. iLBC
    6. GSM

12. Who are ISP and ITSP?

- ISP stands for "Internet Service Provider" who provides you the internet service
  Example: AIRTEL, RELIANCE (Internet Service Providers in India)
- ITSP stands for "Internet Telephony Service Provider" who provides you telephony services over the internet. Internet telephony allows you to call any PSTN or GSM line from an IP network
  Example: ICENET, SATYAM (Internet Telephony Service Providers in India)
- If you are looking only for peer-peer calling then you need to get internet connectivity from the ISP, there is no need of services from ITSP
- If you are looking for Internet Telephony then you need to get the internet connectivity from ISP as well you also need to obtain a SIP telephony account from the ITSP

13. What is use of VLAN Tagging?

- LAN is defined as a single broadcast domain with Standard (IEEE 802.1Q)
- VLAN is Virtual LAN
- If a user broadcasts information on his/ her LAN, the broadcast will be received by every other user on the LAN
- VLANs allow to logically segmenting a LAN into different broadcast domains
- In order to extend VLANs across different switches, a link called ‘trunk’ must interconnect the switches
- Now, when you want traffic from multiple VLANs to be able to cross a link that interconnects two switches, there is a need to configure a VLAN tagging method on the ports that supply the link
- A frame is tagged when it leaves the switch with information about the “VLAN Id” that the frame belongs to

14. What is NAT (Network Address Translation)?

- NAT is the translation of an IP address used within one network to a different IP address known within another network

- One network is designated the inside network and the other is the outside. Typically, a company maps it’s local inside network addresses to one or more global outside IP addresses and un-maps the global IP addresses on incoming packets back into local IP addresses
- This helps ensure security since each outgoing or incoming request must go through a translation process that also offers the opportunity to qualify or authenticate the request or match it to a previous request
- NAT also conserves on the number of global IP addresses that a company needs and it lets the company use a single IP address in its communication with the world

15. What Is STUN?

- STUN stands for Simple Traversal of UDP over NAT
- It is a protocol for assisting devices behind a NAT, firewall or router with their packet routing
- It allows applications to discover the presence and types of NATs and firewalls between them and the public Internet
- It provides applications with the ability to determine the public IP addresses allocated to them by the NAT
- It operates on TCP & UDP port 3478

16. What is RTP?

- RTP stands for Real-Time Transport Protocol
- It is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services
- RTP provides services such as payload type identification, sequence numbering, time stamping and delivery monitoring to real-time applications

17. What is ‘Jitter buffer’?

- Variations in packet arrival time are called Jitter
- It can occur because of network congestion, timing drift, or route changes