"FAQs on SETU VFXTH"

Q1 . Which Internet Telephony (VoIP) Protocol is supported by SETU VFXTH?

Ans. SETU VFXTH supports Session Initiation Protocol V2 to make/receive calls on Internet Network.

Q2. What is SETU VFXTH?

Ans. SETU VFXTH is a Gateway offering Seamless Connectivity between VoIP, FXO & FXS Networks.

It offers:-

- VoIP & FXO Network Connectivity to Traditional PBX *without increasing FXO Ports of PBX*
- PSTN Network Connectivity to an IP PBX

Q3. Which are the various Variants of VFXTH and Interfaces supported?

Ans. There are 10 Variants of VFXTH:-

Configurations	VoIP Channels	FXO Ports	FXS Ports
VFXTH0016	16	0	16
VFXTH0024	24	0	24
VFXTH0032	32	0	32
VFXTH0800	8	8	0
VFXTH1600	16	16	0
VFXTH2400	24	24	0
VFXTH3200	32	32	0
VFXTH0808	16	8	8
VFXTH1212	24	12	12
VFXTH1616	32	16	16

* Total 32 SIP Trunks are supported in all Variants

Q 7. Who are ISP and ITSP?

Ans. ISP stands for "Internet Service Provider" who provides you the internet service.

- Example: AIRTEL, RELIANCE (Internet Service Providers in India).
- 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 account from the ITSP.

Q 8. What is Peer to Peer Calling?

Ans. Peer to Peer Calling is basically IP – to – IP Calling.

IP-to IP calling is possible in the following applications:

1. Calls inside the same network
2. Calls between VLANs
3. Calls between various Networks

Q.9. What is the default IP Address of Ethernet Port of SETU VFXTH?

Ans. The Default IP Address of Ethernet Port of SETU VFXTH is 192.l68.001.136

Q.10. What is the use of the option "Gateway" Mode for SIP Trunk in SETU VFXTH?

Ans. Select SIP Trunk as Gateway, if you want the system to accept the incoming call for any SIP ID received from the same Registrar / Outbound Proxy Server Address that you configure for that particular SIP Trunk

Q.11. Which are the Vocoders supported by SETU VFXTH?

Ans. The Vocoders supported by SETU VFXTH are :

G.729

G.723.1

GSM FR

G.711 (µ-Law)

G.711 (A-Law)

Q.12. What is the use of the Feature "IP Dialing" in SETU VFXTH ?

Ans. SETU VFXTH supports direct dialing of IP Addresses from the source port. To provide IP Dialing Facility to the users, SIP Trunk/Group needs to be configured for IP Dialing.

Whenever an IP address is dialed out from source port of SETU VFXTH, the system does not check the Destination port Determination Method, instead will route the Dialed IP Address through the SIP Trunk/Group you configured for IP Dialing.

Q.13. In SETU VFXTH with FXS Interface, How do I Program the system to enable Internal Calling between the FXS ports?

Ans. To program the FXS ports for Intercom functionality in SETU VFXTH, the simplest way to configure the Outgoing of the FXS is as per Destination Number Based and program the Flexible Numbers of all the FXS in it.

Q.14. In SETU VFXTH what option should I select for Subscriber type in Outgoing call Handling of FXS Port ?

Ans. When SETU VFXTH is interfaced with a service provider server –ITSP, the Matrix ETERNITY IP-PBX, or any other PBX – that supports supplementary services that require dialing of Flash, like Call Hold, Call Toggle, Call Waiting, you must select the Subscriber Type for SETU VFXTH as Below :-

Network : To use only supplementary services supported by the PBX. In this case you can access the Service Provider/PBX features by dialing flash but will not be able to access the local features of SETU VFXTH.

Gateway : To use primarily the supplementary features of SETU VFXTH. You will also be able to access the supplementary services of the service provider which require dialing of flash followed by the access code #4.

Q.14. How to disconnect a Call using Access Code in SETU VFXTH?

Ans. SETU VFXTH enables users to disconnect a current call using the access code #92

Q.15. Is SIP to SIP Routing Possible in SETU VFXTH?

Ans. Yes SIP to SIP Routing is possible, Call on one SIP Trunk can be routed outside via another SIP Trunk.

Q.16. How to default SETU VFXTH?

Ans. You can restore the Factory defaults by selecting the Option “Default System” in System Maintenance Page through the Web Based Programming Software Jeeves. There is no option of Hardware Default.

Q.17. I have VFXTH 0016, I don’t remember the IP address. How can I get to know the IP Address of the Ethernet port of VFXTH?

Ans. Connect an Analog phone with Display and dial 21-#* & go on hook. You can view the Network IP address.

Q.18. How do I change the IP Address of VFXTH using the Analog phone on FXS Port?

Ans. Enter the programming mode #19-1234 and dial 11-IP Address-#*

Q.19. How to Blind transfer Incoming call on one FXS port to another Party’s Number?

Ans. Enable Call Transfer-Blind in the class of service of the FXS.

Blind transfer : You are in speech with party A & Party A wishes to speak to Party B

You dial Fash & then #6 (Blind Call Transfer Access code)

Party A is put on hold and you hear dial tone

Dial the number of Party B

A will be connected to B

You on hook your phone

Q.20. How to do Attended call transfer from an FXS Port of SETU VFXTH?

Ans. Enable Call Transfer- Attended in the class of service of the FXS.

Blind transfer : You are in speech with party A & Party A wishes to speak to Party B

You dial Fash

Party A is put on hold and you hear dial tone

Dial the number of Party B

When Party B answers the call, go On-hook to Transfer

Q.21. Is there any Option that the Incoming call on a Trunk is answered & dial tone is played to the caller, allowing the caller to dial the desired number?

Ans. Yes there is the Option of “Answer the call & Collect Digits” Under Incoming Call Routing for FXO Port of SETU VFXTH, by which the caller on the trunk gets manual dial tone to dial out the desired number . However this Option is not available for Incoming call on SIP Trunk.