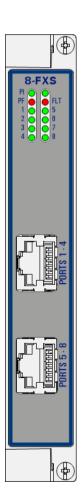


# **User Manual**

Installation Dragon PTN Interface Module PTN-8-FXS



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## 1. INTRODUCTION

## 1.1 General

This document is valid as of Dragon PTN Release 3.0DR.

In general, FXS (=Foreign eXchange Subscriber) and FXO (=Foreign eXchange Office) gateways translate analog voice/modem/fax signals to IP packets and vice versa.

- FXO: towards analog PBX (=Private Branch eXchange) or PSTN;
- FXS: towards analog telephones, modems or faxes.

The 8-FXS interface module (=IFM) is a VoIP module and makes the transport of analog voice through the Dragon PTN packet network possible. It will convert analog voice into SIP (=Session Initiation Protocol) and RTP messages and vice versa. The SIP messages will be packetized into MPLS-TP packets via a Voice service in HiProvision, to transport the data through the Dragon PTN network. HiProvision is the Dragon PTN management system.

The Voice service can set up connections between both analog phones and SIP elements (client, server, trunk). SIP elements can be connected to the Dragon PTN network via Ethernet IFMs (e.g. 4-GC-LW, ...). Depending on where the call handling must be done, two different modes can be selected:

- Remote extension (FXO Gateway): call handling via FXO gateway to analog public or private telephone networks;
- SIP-server: call handling done in dedicated external SIP server;

This IFM provides 8 ports for connecting 8 analog phones to the Dragon PTN network. It can be used in any IFM slot of any node. An IFM slot overview can be found in Ref. [3] in Table 1.

Maximum four Voice services can be configured per 8-FXS module.

The main supported features are:

- Packetizing of analog voice;
- 8 ports FXS (G.712 compliant);
- Integrated battery feed and ringing generator;
- DTMF Dialing;
- Converts analog FXS port into SIP Client;
- Support for G.711 a/μ law and G.729

encoding/decoding; Modem and fax compatible (no

T.38);

- Remote Extension: FXO interworking possible via external device;
- Interworks with standard SIP servers;
- Voice service modes:
  - Remote extension (Routable);
  - SIP-server (Routable);
  - **NOTE:** Routable: FXO Gateway or SIP Server can be located in an IP subnet different from the IP subnet assigned to the 8-FXS ports;
- Compatible products can be found in §3.

A general analog Voice service example can be found in the next figure:

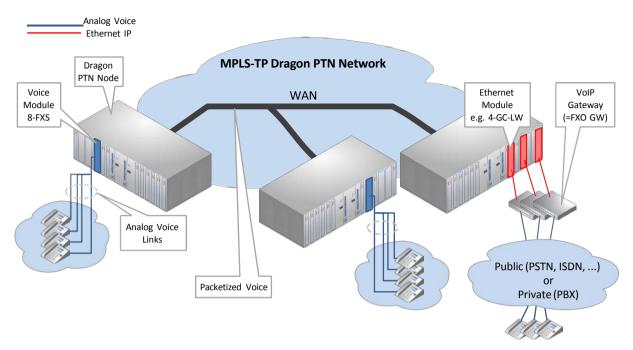


Figure 1 Analog Voice Example: Remote Extension

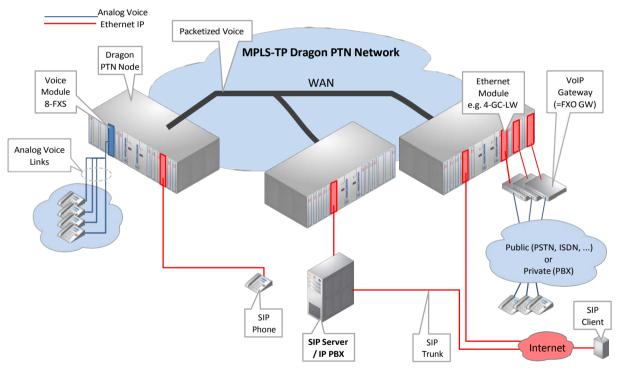


Figure 2 Analog Voice Example: SIP Server

## 1.2 Manual References

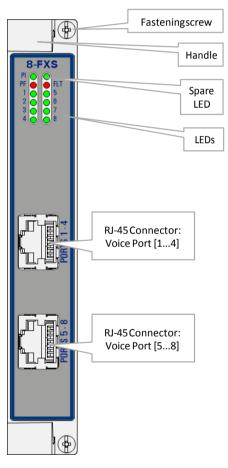
Table 1 is an overview of the manuals referred to in this manual. '&' refers to the language code, '\*' refers to the manual issue. All these manuals can be found in the HiProvision (=Dragon PTN Management System) Help function.

## **Table 1 Manual References**

Ref.	Number	Title
[1]	DRA-DRM821-&-*	Dragon PTN and HiProvision Operation
[2]	DRA-DRM801-&-*	Dragon PTN Installation and Operation
[3]	DRB-DRM802-&-*	Dragon PTN Nodes: PTN2210, PTN2209, PTN2206, PTN1104
[4]	DRB-DRM803-&-*	Dragon PTN Switching Module: PTN-CSM310-A
[5]	DRE-DRM807-&-*	Dragon PTN Interface Module: PTN-4-GC-LW / PTN-4-GCB-LW
[6]	DRE-DRM808-&-*	Dragon PTN Interface Module: PTN-1-10G-LW
[7]	DRE-DRM817-&-*	Dragon PTN Interface Module: PTN-4-GO-LW
[8]	DRA-DRM810-&-*	Dragon PTN General Specifications

## 2. MODULE DESCRIPTION

## 2.1 Front Panel



**Figure 3 Front Panel** 

#### 2.1.1 Handle

#### a. Insert the Module into the Node

Take the front panel handles to insert or slide the module into the Dragon PTN node. Push the module thoroughly into the node's backplane. Next, tighten the two fastening screws in the front panel corners.

## b. Remove the Module from the Node

Untighten the two fastening screws in the front panel corners. Take the front panel handles to pull out and finally remove the module from the Dragon PTN node.

## 2.1.2 LEDs

The meaning of the LEDs depends on the mode of operation (= boot or normal) in which the 8-FXS module currently is running. After plugging in the module or rebooting it, the module turns into the boot operation, see Table 2. After the module has gone through all the cycles in the table below (=rebooted successfully), the module turns into the normal operation, see LEDs in Table 3.

Cycle	PI	PF	FLT	Spare LED	Port[18]
1	х				see Table 3
2	х		Slow blinking		see Table 3
3	x		Fast blinking		see Table 3
4	х				
5	x	x slow blinking			
x : LED is lit / : LED is not lit The sub cycle times may vary. The entire boot cycle time $[1 \rightarrow 5]$ takes approximately 1 minute.					

## Table 2 LED Indications In Boot Operation

## **Table 3 LED Indications In Normal Operation**

LED	Color	Status		
PI (=Power Input)	Not lit, dark	+12V power input to the board not OK		
	Green	+12V power input to the board OK		
PF (=Power Failure)	Not lit, dark	power generation on the board itself is OK		
	Red	power generation on the board itself is erroneous		
FLT (=FauLT)	Not lit, dark	no other fault or error situation, different from PF, is active on the module		
	Red	a fault or error situation, different from PF, is active on the module		
Spare	Not lit, Green	spare		
<port n°=""></port>	Not lit, dark	- port configured (*): connected analog phone is on hook and registered (**)		
	Green, slow blinking	connected analog phone is <u>on hook</u> together with one of the conditions below: - not configured - not registered (**)		
	Green, fast blinking	connected analog phone is <u>off hook</u> together with one of the conditions below: - not configured - not registered (**)		
	Green, lit	connected analog phone is off hook and registered (**)		

(\*) Port Configured: a port is configured when it is programmed in a Voice service and it has a created account. The account must have been created via the Voice Protocol wizard in HiProvision.

(\*\*) Register: The analog phones must have been announced or registered first to a third party FXO gateway or SIP-server. This occurs via a 'REGISTER' message. After registration, the SIP server or FXO gateway knows via which IP address and extension number the analog phones can be reached. The REGISTER message will be sent automatically by the port, every 120 seconds, after the port has been configured and loaded in HiProvision. No phone has to be connected yet to start the registration. See also §2.2.5 for more info on the Register flow.

## 2.1.3 RJ-45 Connector Connections and Cables

The 8-FXS module provides 8 analog voice ports via 2 RJ-45 connectors, each connector providing 4 ports. Each port provides one tip/ring pair. Connections to this IFM can be made via a connection kit that can be plugged as well in a patch panel:

- 8-FXS Connection Kit (future support) includes:
  - An RJ-45 drop cable (1.5m) wired out on 4\*RJ-11 couplers to connect ports [1..4];
  - An RJ-45 drop cable (1.5m) wired out on 4\*RJ-11 couplers to connect ports [5..8];
- Patch panel (order no. 942 256-204):
  - 24 ports feedthrough patch panel. The RJ-45 drop cables from the connection kit can be plugged into this patch panel.
  - Analog telephony RJ11 cables must be plugged into the RJ-11 couplers to connect to the 8-FXS ports;
- See the table and figures below for an overview and description of the different connections, connectors and pin assignments.

## **Table 4 Connection Products**

Description	Ordering Number
8-FXS Connection Kit	future support
Patch panel 1 U feedthrough 24 ports (to be used in combo with the 8-FXS Connection Kit	942 256-204

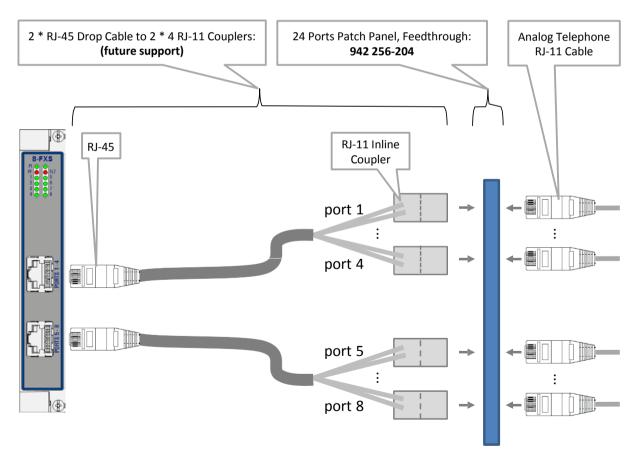


Figure 4 8-FXS Connection: Unplugged

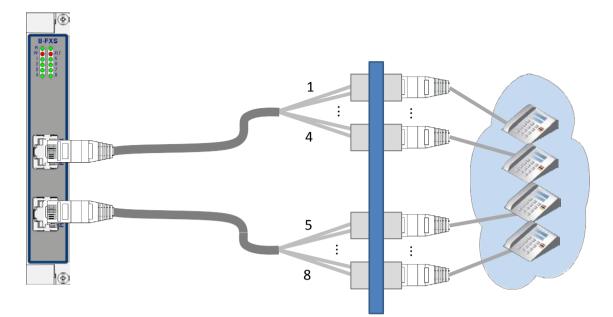


Figure 5 8-FXS Connection: Plugged



Figure 6 8-FXS RJ-45 Connector

8-	FXS RJ4	45 Connect	Coupler			
Port No.	Pin No.	Descript ion	Wire Colors	Coupler No.	Pin No.	Color
1	1	TIP1	white/orange	1	2	Green
	2	RING1	orange		3	White Blue
2	3	TIP2	white/green	2	2	Green
	6	RING2	green		3	White Blue
3	4	TIP3	blue	3	2	Green
	5	RING3	white/blue		3	White Blue
4	7	TIP4	white/brown	4	2	Green
	8	RING4	brown		3	White Blue
5	1	TIP5	white/orange	5	2	Green
	2	RING5	orange		3	White Blue
6	3	TIP6	white/green	6	2	Green
	6	RING6	green		3	White Blue
7	4	TIP7	blue	7	2	Green
	5	RING7	white/blue		3	White Blue
8	7	TIP8	white/brown	8	2	Green
	8	RING8	brown		3	White Blue

## Table 5 8-FXS RJ-45 Cable-Coupler: Pin Assignments

## 2.2 Functional Operation

## 2.2.1 General

An application network can be connected to the MPLS-TP Dragon PTN network via one of the 8 FXS interface ports. The 8-FXS module can interface with 8 analog telephone lines. In Figure 1 and Figure 2, a common functional setup is shown.

In Figure 7 below, a more detailed functional setup is shown. An application network interfaces the Dragon PTN node via the FXS ports on the 8-FXS module. The 8-FXS converts this traffic into Ethernet traffic on the backplane. The Central Switching Module (=CSM310-A) converts this Ethernet traffic into packetized MPLS-TP and transmits it via an Ethernet IFM (e.g. 4-GC-LW) onto the Dragon PTN MPLS-TP network.

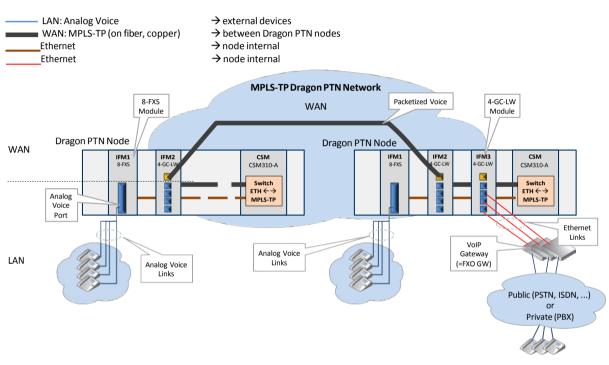


Figure 7 Detailed Functional 8-FXS Example

## 2.2.2 I/O with the Central Switching Module (=CSM)

The 8-FXS module receives analog voice via its front panel ports and converts this into Ethernet traffic which is forwarded to the CSM via the backplane. The CSM does all the processing on this data (synchronization, CRC checks, conversions, switching...). The CSM converts this data into MPLS-TP packets and transmits it via an Ethernet IFM (e.g. 4-GC-LW) onto the WAN. On the destination side, the same processing occurs in reverse order.

## 2.2.3 Voice Service: Remote Extension (FXO Gateway)

- See Figure 1: connections between FXS  $\leftarrow \rightarrow$  VoIP Gateway (=FXO-GW);
- Analog phones in the Dragon PTN network are a remote extension of the public (PSTN) and/or private (PBX) telephone network. The extension is possible via a third party FXO-GW (e.g. Patton);

- Analog phones are connected to an FXS interface (8-FXS module);
- FXO Gateways are connected to an Ethernet interface (e.g. 4-GC-LW module);
- Voice switching and telephony feature handling between all phones are performed in the PSTN or PBX;
- DTMF (=Dual Tone Multi-Frequency) must be used for number dialing;
- Analog speech and signaling are converted and packetized by the FXS interface and the FXO-GW towards the Dragon PTN network

To set up an entire Voice service in HiProvision (See Ref.[1] in Table 1):

- a. Configure a Voice service in HiProvision. A voice service requires at least one 8-FXS port and one Ethernet port (e.g. 4-GC-LW).
- b. Configure a Voice Protocol in the configured Voice service. This voice protocol is necessary to configure extra/enhanced voice parameters.
- **NOTE:** 8-FXS ports are always consumed per pair in a Voice service: port1+port2; port3+port4; port5+port6; port7+port8. So if you configure port5 in a first service e.g. Voice1, port6 can only be used in the same Voice1 service.

## 2.2.4 Voice Service: SIP-Server

- See Figure 2: SIP elements like a SIP Phone and VoIP gateway can be registered on and handled by the SIP Server;
- Analog phones are connected to an FXS interface (8-FXS module) which are registered on the SIP Server;
- Voice switching and telephony feature handling for all calls are performed in the SIP Server;
- Possible SIP telephony features depend on the used SIP devices and their interoperability.

To set up an entire Voice service in HiProvision (See Ref.[1] in Table 1):

- c. Configure a Voice service in HiProvision. A voice service requires at least one 8-FXS port and one Ethernet port (e.g. 4-GC-LW).
- d. Configure a Voice Protocol in the configured Voice service. This voice protocol is necessary to configure extra/enhanced voice parameters.
- **NOTE:** 8-FXS ports are always consumed per pair in a Voice service: port1+port2; port3+port4; port5+port6; port7+port8. So if you configure port5 in a first service e.g. Voice1, port6 can only be used in the same Voice1 service.

#### 2.2.5 SIP Connections

After a Voice service and Voice protocol has been configured in HiProvision, voice calls (or connections) between analog and/or SIP clients can be set up via SIP calls.

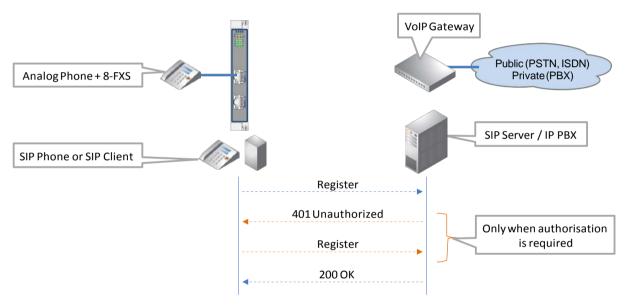
Before any SIP call is possible between end clients A and B, both A and B must register first to a SIP Server, whatever mode is used, either Remote Extension or SIP Server. A registration

attempt from the 8-FXS IFM automatically occurs (every 120 seconds) when the port on which A or B is or will be connected, is configured in HiProvision. As a result of a successful registration process, A and B and their location will be known by the SIP server.

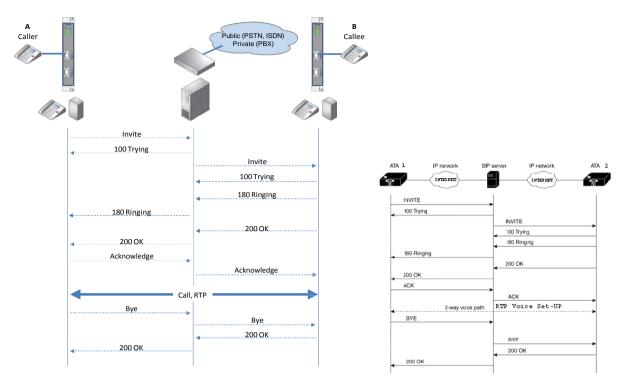
- Remote Extension mode: The external FXO Gateway device acts as SIP server;
- SIP Server mode: A Server PC with SIP Server software can act as SIP server;

Once the registration has been done, a voice call between A and B is possible. Basically, when A calls B, a SIP Invite message will be sent to the SIP server first, which redirects the Invite message to B. B starts ringing, picks up the phone and the Voice call has been set up. Once the voice call has been established, Real-time Transport Protocol (=RTP) is used to transport the call data.

- Remote Extension mode: the RTP stream always goes through the FXO Gateway device;
- SIP-Server mode: It can be configured in the SIP Server itself whether the RTP stream goes through the SIP Server. If for example the calls must be monitored or recorded, the RTP must go through the SIP server. Find below some example SIP flow examples.



**Figure 8 Basic SIP Registration Process** 



## Figure 9 Basic SIP Call Example Setup Between End-Points

CAUTION: The call setup in the figure above may slightly differ depending on the selected mode!

## 2.3 Onboard Interfaces



#### Figure 10 8-FXS: Side View

## 2.3.1 Straps

No user relevant straps.

## 2.3.2 DIP Switches

No user DIP switches.

## 2.3.3 Hardware Edition

The hardware edition of the IFM has been factory set and can not be changed. It can be read out via HiProvision in the Software tile, see Ref. [1] in Table 1.

## 3. COMPATIBILITY

The 8-FXS IFM is compatible with:

- ▶ 4-GC-LW IFM, 4-GCB-LW IFM, 1-10G-LW IFM, 4-DSL-LW IFM;
- ► FXO Gateway: Patton SmartNode: SN4332/JO/UI (Multiport Analog Voip gateway): SmartNode Multiport FXO VoIP GW, 32 FXO ports presented on RJ21, 1x Fast Ethernet, Internal UI Power. NOTE: SN43xx : → xx indicates amount of ports. Other similar Patton devices with more/less ports are supported as well.

It means that 8-FXS and all the products mentioned above can be programmed in the same Voice service.

## 4. MODULE SPECIFICATIONS

#### 4.1 General Specifications

For general specifications like temperature, humidity, EMI ... see Ref.[8] in Table 1.

## 4.2 Other Specifications

## **Table 6 Other Specifications**

Description	Value		
Weight	0.23 kg / 0.5 lb		
MTBF	185 years at 25°C/77°F		
Power Consumption	Measured at 25°C/77°F, with data transport: - All phones on-hook: 5.6W - Extra per phone off-hook: 0.9W - Ringing state: max. 9W - All phones off-hook: 12.8W		
Module Size	width: 20.32 mm / 0.8 inches   height: 126 mm / 4.96 inches   depth: 195 mm / 7.68 inches		

## 4.3 Ordering Information

- PTN-8-FXS: future support;
- PTN-8-FXS Connection Kit: future support;
- Patch panel 1 U feedthrough 24 ports (to be used in combo with the 8-FXS Connection Kit): 942 256-204.

#### 5. ABBREVIATIONS

CSM	Central Switching Module
DTMF	Dual Tone Multi-Frequency
EMI	Electromagnetic Interference
ETH	Ethernet
FLT	Fault
FXO	Foreign eXchange Office
FXS	Foreign eXchange Subscriber
GW	Gateway
IEEE	Institute of Electrical and Electronics Engineers
IFM	InterFace Module
IP	Internet Protocol
ISDN	Integrated Services Digital Network
kbps	Kilobit per Second
LAN	Local Area Network
LOS	Loss Of Signal
Mbps	Megabit per Second
MPLS-TP	MultiProtocol Label Switching – Transport Profile
MTBF	Mean Time Between Failures
PBX	Private Branch Exchange
PF	Power Failure
PI	Power Input
PTN	Packet Transport Network
PSTN	Public Switched Telephone Network
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
VoIP	Voice over IP
WAN	Wide Area Network