



**swisscom**

# Audiocodes MediaPack 5xx (Formerly Mediant 500Li)

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SIP Trunk Configuration Guide for Enterprise SIP



**From** Enterprise SIP Engineer  
**Date**  
**Subject** SIP Trunk Configuration Guide for Enterprise SIP

**To** IP PBX and  
Communication System  
Integrators

**For information**

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## 1 Introduction

### 1.1 Objective and purpose

Describes the SIP Trunk configuration of IP PBX or communication systems. The IP PBX or communications systems are homologated using this SIP Trunk configuration in order to interoperate with the Enterprise SIP service.

More information about the Enterprise SIP service is available here: <https://www.swisscom.ch/de/business/enterprise/angebot/wireline-networks-bundles/business-voice/sip-trunking.html>

### 1.2 Target audience, requirements for the reader

IP PBX and Communication System Integrators.

### 1.3 Terms, abbreviations

Terms	Abbreviations
ATA	Analog Telephone Adapter, used to integrate analog endpoints into VoIP systems
eSBC	SBC build into the Swisscom Cisco router
LIS	Location Information Service of Skype for Business
PSTN	Public Switched Telephone Network
SBC	Session Boarder Controller, device for termination of voice over IP networks. It controls signalization as well as media streams.

### 1.4 Referenced documents

[1] SIP Trunk Technical User Network Interface Guide for Enterprise SIP (UNI Guide)

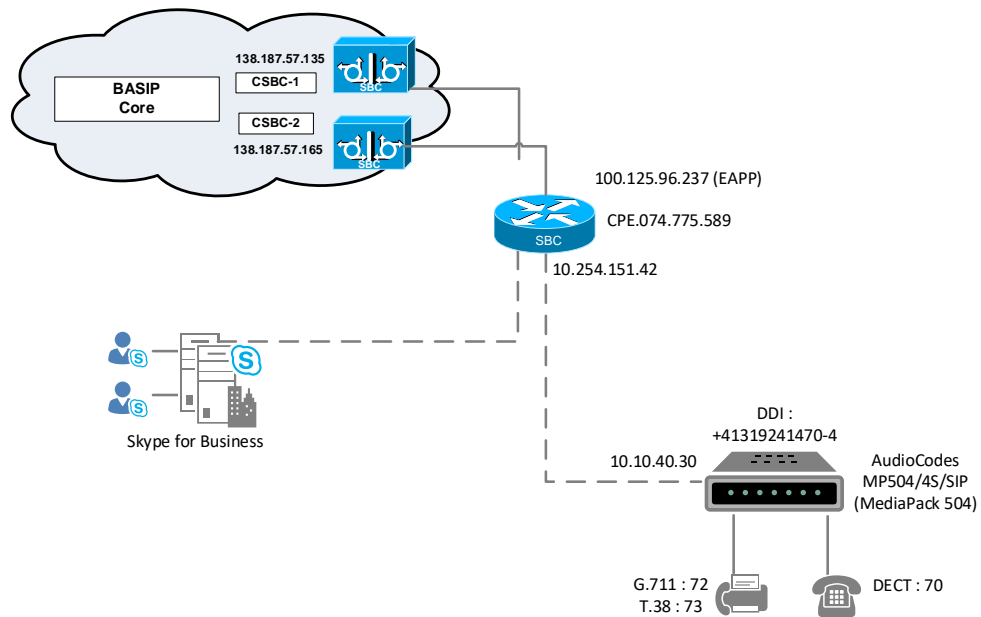
## 2 Overview

This document describes only the configuration parameters for Audiocodes MediaPack 5xx (formerly Audiocodes Mediant 500L-i, rebranding done by Mid of April 2023)

Only the Standard Enterprise SIP trunk (without redundancy) is described in detail. For high availability and disaster recovery scenario please contact Swisscom Spec Sales and Design team over your account manager to design an appropriate system architecture to match the customers' need.

### 2.1 SIP Trunk network architecture

The Enterprise SIP Trunk is terminated on the eSBC, located on of the Cisco router.



### 2.2 Hardware requirements

This Audiocodes device can be used in conjunction with Skype for Business Server 2015 and Lync Server 2013. Please take the configuration's parameters for Skype for Business Server 2015 and Lync Server 2013 from the respective homologations.

### **2.3 Software requirements**

The homologation was made with this version :

- Audiocodes MediaPack 504 (MP504) version 7.26A.356.174

**Note:**

Older product version and builds were not tested and may not work properly

### **3 SIP Trunk features**

#### **3.1 Features supported and tested**

- National calls
- International calls
- National calls with international prefix
- Toll free numbers (0800)
- DTMF (RFC 2833, default Payload Type 101)
- Call cancellation
- Call rejection
- Calls with early media
- Calls to special/short numbers
- Calling line indication presentation (CLIP)
- Calling line indication restriction (CLIR)
- Special Arrangement
- Call hold/resume
- Music on hold
- Call forwarding unconditional
- Call forwarding busy
- Call forwarding no answer
- Attended call transfer
- Blind call transfer
- 3-party conference
- T.38 Fax
- Fax with G.711 pass-through
- Keep alive out of session

#### **3.2 Caveats and known restrictions**

- Modem not tested
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number

##### **3.2.1 Features not supported**

- E112 with geolocation. Instead, the geolocation is based on A-number and performed in SIP Core.



#### 4 SIP Trunk configuration Audiocodes MediaPack 5xx (MP5xx)

The below settings are made upon the factory reset of the box. They ensure that the full test plan will be successful. As soon as a setting differs from the default, it is visible as below:



The IP network part as well as the core entities of the Signaling & Media are not described in the current document. For Basic configuration please refer to the [MediaPack 5xx Voice Gateway Basic Configuration](#) document.

##### 4.1 General settings

Signaling & Media->SIP Definitions->Message Structure

Parameter	Value
Asserted Identity Mode	Add P-Asserted Identity

Signaling & Media->SIP Definitions->Transport Settings

Parameter	Value
SIP Transport Type	TCP

Signaling & Media->SIP Definitions-> Proxy & Registration

Parameter	Value
Use Gateway Name for OPTIONS	Server
Registrar Transport Type	TCP
Always Use Proxy	Enable

Signaling & Media->Gateway-> DTMF & Supplementary->DTMF & Dialing

Parameter	Value
Max Digits In Phone Num	25

Signaling & Media->MediaRTP/RTCP Settings

Parameter	Value
Comfort Noise Generation Negotiation	Disable

Signaling & Media->Message Manipulation->Message Manipulations

Manipulation Set ID 0	
Parameter	Value
Message Type	Invite.Response.3xx
Condition	Header.Diversion.URL.Type == '2'
Action Subject	header.diversion.url.type
Action Type	Modify
Action Value	'1'

Signaling & Media->Message Manipulation->Message Manipulations

Manipulation Set ID 0	
Parameter	Value

Message Type	Invite.Response.3xx
Condition	Header.Diversion.URL.Type == '2'
Action Subject	Header.Diversion.URL.Host.Name
Action Type	Add
Action Value	'10.10.40.30' (pls replace with the IP address of the LAN interface)

Signaling & Media->Core Entities->IP Groups (in addition to the necessary, but not described parameters)

Name ESIP	
Parameter	Value
Outbound Message Manipulation	0

#### 4.2 Manipulation & Routing

Signaling & Media->Gateway->Manipulation->Dest Number Tel->IP

Rule E164_INTERNAT	
Parameter	Value
Destination Prefix	00*
Destination IP Group	ESIP
Stripped Digits From Left	2
Prefix to Add	+

Rule E164_NATIONAL	
Parameter	Value
Destination Prefix	0[1-9]xxxxxxxx
Destination IP Group	ESIP
Stripped Digits From Left	1
Prefix to Add	+41

Rule E164_COMBOX	
Parameter	Value
Destination Prefix	0860xxxxxxxx
Destination IP Group	ESIP
Stripped Digits From Left	1
Prefix to Add	+41

Rule E164_EMERGENCY	
Parameter	Value
Destination Prefix	11x, 144, 145, 141x
Destination IP Group	ESIP
Stripped Digits From Left	0
Prefix to Add	+41

Rule E164_SERVICE	
-------------------	--

Parameter	Value
Destination Prefix	1x*
Destination IP Group	ESIP
Stripped Digits From Left	0
Prefix to Add	+41

Optional:

Rule E164_GEOLOC	
Parameter	Value
Destination Prefix	086756
Destination IP Group	ESIP
Stripped Digits From Left	1
Prefix to Add	+41

Signaling & Media->Gateway->Source Number IP->Tel

Rule EXPL_to_IMPL_+41	
Parameter	Value
Source Phone Pattern	+41*
Source IP Group	ESIP
Stripped Digits From Left	3
Prefix to Add	0

Rule EXPL_to_IMPL	
Parameter	Value
Source Phone Pattern	+*
Source IP Group	ESIP
Stripped Digits From Left	1
Prefix to Add	00

Signaling & Media->Gateway->Routing->Tel->IP Routing

Rule FXS_to_ESIP	
Parameter	Value
Destination IP Group	ESIP

Signaling & Media->Gateway->Routing->IP->Tel Routing

Rule ESIP_to_FXS	
Parameter	Value
Source SIP Interface	any
Destination Phone Pattern	+4131924147[0-4]
Trunk Group ID	1
Source IP Group	ESIP
Trunk ID	1

Signaling & Media->Gateway->Trunk Group Settings

Name by_number	
Parameter	Value
Trunk Group ID	1
Channel Select Mode	By Dest Phone Number
Used By Routing Server	Yes
Serving IP Groupe	ESIP

### 4.3 Coder & Profiles

Signaling & Media->Coders and Profiles->Tel Profile

Parameter	Value
Profile ID	1
Profile Name	G.711 only
Fax Signalling Method	No Fax
Progress Indicator to PI	No PI
Echo Canceler	Disable
Coder Group	AudioCoderGroups_0

Optional: Tel Profile for T.38: not necessary as default configured behavior is T.38 relay

Parameter	Value
Profile ID	2
Profile Name	T.38
Fax Signalling Method	T.38 Relay

### 4.4 Fax part

Signaling & Media->Media->Fax/Mode/CID Setting

Parameter	Value
Fax Transport Mode	Disable
V.22 Modem Transport Type	Disable
V.23 Modem Transport Type	Disable
V.32 Modem Transport Type	Disable
V.34 Modem Transport Type	Disable
Fax Relay Max Rate (bps)	9600bps
Fax Relay ECM Enable	Disable
Fax CNG Mode	Send on CNG tone

Signaling & Media->Gateway->Gateway General Settings

Parameter	Value
Fax Signalling Method	Fax Fallback

SIP T.38 Version	Version 0
Channel Select Mode	By Dest Phone Number

Optional: To test T.38 with sending fax sending T.38 reINVITE immediately after establishment of the call:  
Signaling & Media->Media->Fax/Mode/CID Setting

Parameter	Value
CNG Detector Mode	Relay instead of Disable

#### 4.5 Voice part

Signaling & Media->Media->Fax/Mode/CID Setting

Parameter	Value
Caller ID Transport Type	Relay

Signaling & Media->Gateway->DTMF & Supplementary->Supplementary Services Settings

Parameter	Value
Enable Caller ID	Enable

#### 4.6 Trunk & Number assignment (example)

Signaling & Media->Gateway->Trunk Groups

(In our configuration, +41319241470 & 71 are reserved for DECT devices, 72 for G.711 only fax and 73 for T.38 fax)

GROUP INDEX	MODULE	FROM TRUNK	TO TRUNK	CHANNELS	PHONE NUMBER	TRUNK GROUP ID	TEL PROFILE NAME
1	Module 2 FXS			1	+41319241470	1	None
2	Module 2 FXS			2	+41319241471	1	None
3	Module 2 FXS			3	+41319241472	1	G.711 only
4	Module 2 FXS			4	+41319241473	1	None

#### 4.7 Optional customer defined settings

The below settings are not mandatory and can be freely defined by the customer. The below examples are taken from the homologation's testing environment.

##### 4.7.1 Keypad Features

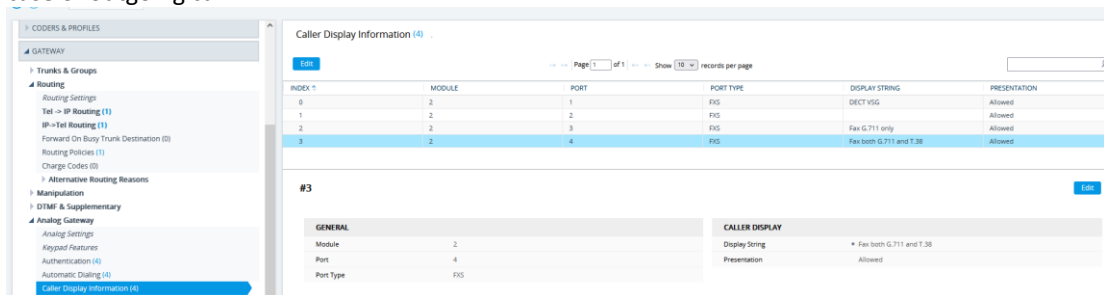
Signaling & Media->Gateway->Analog Gateway->Keypad Features

Parameter	Value
Hook-Flash Code	*56
Forward Unconditional	*21
Forward No Answer	*61
Forward On Busy	*67
Forward Deactivate	#20
Restricted Caller ID Activate	*98

Restricted Caller ID Deactivate	#98
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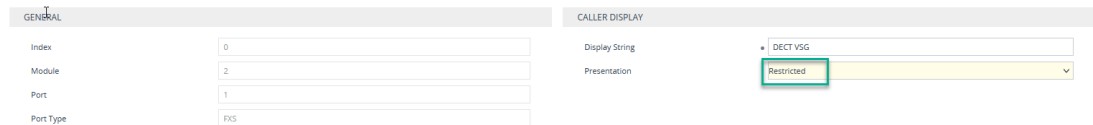
#### 4.7.2 Caller Display Information

- Name of the FXS subscriber, i.e., the name which will be sent in the FROM field beside of the number in case of outgoing call.



INDEX	MODULE	PORT	PORT TYPE	DISPLAY STRING	PRESENTATION
0	2	1	FXS	DECT VSG	Allowed
1	2	2	FXS		Allowed
2	2	3	FXS	Fax G.711 only	Allowed
3	2	4	FXS	Fax both G.711 and T.38	Allowed

- In case of outgoing call and the presentation of the Caller ID must be restricted, the configuration can be achieved either over the web GUI or with the keypad feature:



GENERAL		CALLER DISPLAY	
Index	0	Display String	DECT VSG
Module	2	Presentation	Restricted
Port	1		
Port Type	FXS		

Procedure on the FXS subscriber to activate the Restricted Caller ID:

\*98 <hear the confirmation tone>

Procedure on the FXS subscriber to deactivate the Restricted Caller ID:

#98 <hear the confirmation tone>

#### 4.7.3 Caller ID Permissions

In case of incoming call, the caller ID is per default displayed on the FXS subscriber. This setting is enabled for all the ports, but it is possible to disable this feature per FXS subscriber:

Caller ID Permissions (4)

[Edit](#) Page 1 of 1 Show 10 records per page

INDEX	MODULE	PORT	PORT TYPE	CALLER ID
0	2	1	FXS	Disable
1	2	2	FXS	
2	2	3	FXS	
3	2	4	FXS	

#0

GENERAL		CALLER ID	
Module	2	Caller ID	• Disable
Port	1		
Port Type	FXS		

#### 4.7.4 Call Waiting

Signaling & Media->Gateway->DTMF & Supplementary->Supplementary Services Settings

Parameter	Value
Enable Call Waiting	Disable

Call Waiting (4)

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INDEX	MODULE	PORT	PORT TYPE	ENABLE CALL WAITING
0	2	1	FXS	Enable
1	2	2	FXS	Disable
2	2	3	FXS	Disable
3	2	4	FXS	Disable

#### 4.7.5 Call Forward

Call Forward (4)

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INDEX	MODULE	PORT	PORT TYPE	TYPE	FORWARD DESTINATION	NO REPLY TIME
0	2	1	FXS	Unconditional	0319241421	30
1	2	2	FXS	Deactivate		30
2	2	3	FXS	Deactivate		30
3	2	4	FXS	Deactivate		30

Procedure on the FXS subscriber to activate an unconditional forward to 0319241421:

\*21 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to activate a forward on no answer to 0319241421:

\*61 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to activate a forward on busy to 0319241421:

\*67 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to deactivate any kind of forward:

#20 <hear the confirmation tone>

For the forward on no answer, the timer must be adjusted, if needed, over the web GUI:

GENERAL		CALL FORWARD	
Index	<input type="text" value="0"/>	Forward Destination	<input type="text" value="* 0319241421"/>
Module	<input type="text" value="2"/>	No Reply Time	<input type="text" value="10"/>
Port	<input type="text" value="1"/>		
Port Type	<input type="text" value="FXS"/>		
Type	<input type="text" value="No answer"/>		

#### 4.7.6 Alternative number for outgoing calls

It is possible to use another number than the regular number for outgoing call. To achieve that, a source number manipulation must be created.

Example to use the alternative number +41800800800 instead of the regular number +41319241470:

*Source Phone Number Manipulation for Tel-to-IP Calls*

GENERAL		ACTION	
Index	<input type="text" value="0"/>	Stripped Digits From Left	<input type="text" value="0"/>
Name	<input type="text"/>	Stripped Digits From Right	<input type="text" value="0"/>
		Number of Digits to Leave	<input type="text" value="0"/>
		Prefix to Add	<input type="text" value="*41800800800"/>
		Suffix to Add	<input type="text"/>
		TON	<input type="text"/>
		NPI	<input type="text"/>
		Presentation	<input type="text"/>

MATCH	
Source Trunk Group	<input type="text" value="-1"/>
Source Phone Pattern	<input type="text" value="*41319241470"/>
Destination Phone Pattern	<input type="text" value="*"/>

Cancel **APPLY**



## 5 SIP Trunk requirements and recommendations

The Swisscom requirements described in this chapter are a referencing the SIP Trunk UNI Guide for Enterprise SIP [1].

### 5.1 URI & UIFN Addressing

A number is a string of decimal digits that uniquely indicates the network termination point (hereafter called dial string). The dial string contains the information necessary to route the call to this point. Over-decadic digits are NOT supported. A dial string MUST therefore include only the following characters: 0-9,\*,#,+ ('#' character is represented as "%23", based on RFC 2396). Dial string is expressed in the User-URI Portion of the SIP URI according to RFC 3966. The number will be transported in the user part of the URI.

A name is a string of characters acting as a more readable alias for a given number. It MAY or MAY NOT be presented to the terminating user depending on the target system. Characters used for Names Strings SHALL use the ISO 10646 character set in UTF-8 encoding (RFC 2279). The name will be transported in the display part of the URI.

#### 5.1.1 SIP URI Addressing scheme

Numbering and Addressing Scheme used in all SIP Headers MUST be E.164-based and follow the format defined in RFC3966 Chapter 5.1.4 Global Number format. These numbers MUST be expressed using SIP URI. TEL URI is NOT supported in any header.

In order to successfully setup a call, all telephone number used in the SIP URI MUST be a valid E.164 number preceded with the "+" character and the *user* parameter value "**phone**" MAY be present as described in RFC 3261 section 19.1.1.

In Summary:

- Standard Swiss Geo- and None-Geo Numbers MUST be prefixed by **+41** (e.g., +41582219911)
- Domestic Toll-Free, Shared Cost and Premium MUST be prefixed by **+41** (e.g., +41800800800)
- International Numbers MUST be prefixed by the corresponding Country Code [**+CC**] (e.g., +33142764040)
- International Toll-Free MUST be prefixed by the corresponding Country Code [**+CC**] (e.g., +44800800800)
- Universal International Free phone Number (UIFN) MUST be prefixed by **+** (e.g., +80012345678)
- All 3- and 4 digits Short-Numbers MUST be prefixed by **+41** (e.g., +41117, +411414)

All other formats are NOT supported (i.e., national, subscriber, private, etc.)

The above Requirements SHALL be applied to ALL SIP Headers containing URI Information but MUST be applied at least to the R-URI, FROM, TO, PAI, History-Info and Diversion Headers. If not properly applied on Optional Headers such as P-Preferred-Identity, proper CLID Propagation and/or per-subscriber Billing MAY not be possible. This requirement is valid for SIP requests and responses (i.e., 200OK or 181CallsBeingForwarded).

When sending a INVITE Request, the UA SHALL ensure that Request-URI set to the same URI value of the To header. Examples of a **valid** SIP URI are:

<sip:+41582219911@xxx.xxx.xxx.xxx> <sup>1</sup>

<sip:+41582219911@xxx.xxx.xxx.xxx;user=phone> <sup>1</sup>

<sip:+41582219911@domain.com> <sup>2</sup>

<sip:+41582219911@domain.com;user=phone> <sup>2</sup>

<sip:anonymous@anonymous.invalid> <sup>3</sup>

Examples of **invalid** SIP URI are:

<sip:0582219911@domain.com;user=phone>  
<sip:0041582219911@domain.com;user=phone>  
<sip:41582219911@domain.com;user=phone>  
<sip:@domain.com>

<sup>1</sup> xxx.xxx.xxx.xxx = represents a valid IPv4 Address and SHALL be the public IP-Address of the Originating PBX/SBC in case of FROM/PAI/DIV/Hi and sent-to IP-Address in the R-URI and TO Header.

<sup>2</sup> *domain.com* represents either the domain name that matches the corresponding SIP domain of the originating or the terminating Network (depending on Header) and is subject of bilateral peering Partners. In PAI Header, *domain.com* SHALL reflect the origin Domain of the calling-party if applicable.

<sup>3</sup> applies to the FROM Header ONLY in case of CLIR.

## **5.2 Out of the session keepalive: SIP OPTIONS or INFO**

SIP OPTIONS, respectively INFO, MUST be supported and accepted by the UA. The UA MUST respond with a *200OK* or *483TooManyHops* in case the Hop Count has been set to 0 as done by the Swisscom Border Controller.

For sending SIP OPTIONS or INFO, if possible and/or supported by the UA, the Max-Forwards Header SHALL be set to 0 to terminate SIP Probing right at the ingress point. For all other SIP Messages, the Max-Forwards Header SHALL be left to the default value (70) as defined in Section 11 of RFC 3261.

Since the SIP OPTIONS or INFO are intended as a probing mechanism for SIP reachability between Swisscom border element and UA it is recommended that the URI of the request is ignored (as example UA should not attempt to identify a local user matching the R-URI). SIP OPTIONS SHALL be addressed to the sent-to IP-Address.

The UA MAY provide in the SIP response to OPTIONS or INFO request his capabilities (Allow and Supported header as well SDP including all possible media capabilities). Currently these capabilities are not actively used in call processing but may be part of future extensions.



**Swisscom recommendation:**

To speed up failover in redundant environments (with Multi SAP, SDT-1 or Multi Server Enterprise SIP service options), it is recommended to set the SIP OPTIONS timers to the following values:

- Up-Interval 10sec
- Down-Interval 10sec
- Retry count 2

### **5.3 DTMF**

#### **5.3.1 Sending (to Swisscom)**

DTMF Signals SHALL be sent according to the IETF RFC's 2833/4733. SIP INFO is currently NOT supported. In cases where DTMF Tones are sent in-band in a G.711 RTP Stream, it is transparent to the Network and proper DTMF transmission across the Network can therefore not be guaranteed by either side.

In case of DTMF transmission the SDP MUST contain the *rtptime* and *fmtp* attributes associated with the DTMF payload.

**Swisscom recommendation:**

DTMF Signals sent according to the IETF RFC's 2833/4733 offer the best compatibility with most systems.

#### **5.3.2 Receiving (from Swisscom)**

To insure (backward) compatibility with a system that doesn't support/send out-of-band DTMF (RFC 2833), a system MUST be capable to accept both in-band DTMF (G.711 payload) and out-of-band DTMF (RFC's 2833/4733)

**Swisscom recommendation:**

If the system is depending on DTMF Signals (e.g., Contact Center, Voicemail, etc.) and is not capable to handle both DTMF methods, it is in the responsibility of the solution provider to install appropriate equipment to convert between the two signaling methods.