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3. RESULTS AND DISCUSSION

3.1. Convergent validity and reliability

To guarantee that the "factor loading and average variance extracted (AVE) are higher than 0.70 and 0.50 to fully fulfill the criteria of acceptance," as indicated in previous studies [33], [34]. During the development of the measurement model, the model's convergent validity and reliability were first evaluated. Table 1 demonstrates that this requirement is met by all pertinent values.

3.2. Discriminant validity

The results of the variable's correlation, which was used to assess the measurement's discriminant validity, are displayed in Table 2. To get to the stage of convergent validity examination, the average values must be at least 0.50 AVE and the factor loads must be equal to or greater than 0.70. According to a study by Henseler *et al.* [35], Table 2 shows that every latent construct measurement was completely discriminatory.

3.3. Hypothesis

The hypothesis test indicates that all hypotheses have a favorable direct effect and are therefore accepted, as shown in Table 3. The p-value for ELS and the SVQ composite reliability (CR) value were 7.722 and 0.000, respectively. The p-value of ELS and the ICS CR value were 9.881 and 0.000, respectively. Thus, H1 and H2 were therefore supported.

3.4. Discussion

The analysis's results show a positive relationship between internet access and ELS that is positive, supporting hypothesis H1. This conclusion is consistent with the previous research findings [26], [36]. Numerous studies have shown that learners' satisfaction with their e-learning experience is significantly influenced by ICS [26], [36]–[44]. As has been shown, an ICS is crucial to ELS, which is a critical metric of a system's effectiveness.

The research supports hypothesis H2 by supporting SVQ and ELS. This result agrees with the previous studies [20], [45]. Various research investigations have demonstrated that the quality of service has a major impact on learners' satisfaction with their e-learning experience [20], [45]–[53]. The significant effects of service dimensions on satisfaction thus imply that these elements are crucial for obtaining satisfaction, better value, and competitiveness [54]. Finally, for practical contribution, this study offers suggestions on how educational establishments might successfully integrate the e-learning system to help students with their education.

Table 1. Results of convergent validity and reliability

Constructs	Items	Loading	CR	AVE
ICS	ICS1	0.805	0.999	0.640
	ICS2	0.805		
	ICS3	0.770		
	ICS4	0.811		
	ICS5	0.811		
SVQ	SVQ1	0.783	0.887	0.612
	SVQ2	0.772		
	SVQ3	0.724		
	SVQ4	0.814		
	SVQ5	0.815		
ELS	ELS1	0.794	0.898	0.596
	ELS2	0.781		
	ELS3	0.742		
	ELS4	0.769		
	ELS5	0.768		
	ELS6	0.775		

Table 2. Results of decrement validity

Constructs	ICS	SVQ	ELS
ICS	0.800		
SVQ	0.492	0.782	
ELS	0.738	0.689	0.772

Table 3. Hypotheses result

H	Path	Estimates	St. deviation	CR	R-value	Decision
H1	SVQ→ELS	0.418	0.052	7.722	0.000	Supported
H2	ICS→ELS	0.572	0.049	9.881	0.000	Supported

Factors effects on e-learners' satisfaction at the higher education institutions in Saudi Arabia (Atteeq Ahmad)

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Cisco Implementing Cisco Advanced Call Control and Mobility Services Sample Questions (Q175-Q180):

NEW QUESTION # 175

The Cisco Unified Communications Manager Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

- A. device pools
- B. IP phones
- C. CTI ports
- D. translation patterns
- E. CTI route points

Answer: B,C

Explanation:

Section: Call Control and Dial Planning

Explanation/Reference:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/11_5_1/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115_chapter_01.htm#CUCM_TP_A5DA99E0_00

NEW QUESTION # 176

Refer to the exhibit. An administrator is trying to test outbound calls toward the ITSP but cannot complete the call and receives a SIP error. ITSP is consulted, and the issue is that the ANI that is being sent is not the DID provided 8005532447. Which configuration change sends the correct ANI on the INVITE sent to ITSP to fix the error?

```

Sent:
INVITE sip:9011919999321453@cisco.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.14:5060;branch=z9hG4bKFE45715AD
Remote-Party-ID: "User A" <sip:4444@cisco.com>;party=calling;screen=no;privacy=off
From: "User A" <sip:4444@cisco.com>;tag=E141986A-D65
To: <sip:9011919999321453@cisco.com>
Date: Thu, 31 Jan 2019 19:34:47 GMT
Call-ID: 1A0EC126-24C611E9-81D8D031-86D18CDB@cisco.com
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 1127492352-0000065536-0000014911-0203951114
User-Agent: Cisco-SIPGateway/IOS-15.4.3.S7
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1548963287
Contact: <sip:4444@192.168.1.14:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 668

v=0
o=CiscoSystemsCCM-SIP 113060473 1 IN IP4 192.168.1.16
s=SIP Call
c=IN IP4 192.168.1.14
b=TIAS:384000
b=AS:384
t=0 0
m=audio 24500 RTP/AVP 9 0 8 116 18 101
-Output omitted

007008: Jan 31 19:34:47.246: //1655739/43342800000/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 403 From: URI not recognized
Via: SIP/2.0/UDP 192.168.1.14:5060;branch=z9hG4bKFE45715AD
From: "User A" <sip:4444@192.168.2.1>;tag=E141986A-D65
To: <sip:9011919999321453@cisco.com>;

Call-ID: 1A0EC126-24C611E9-81D8D031-86D18CDB@cisco.com
CSeq: 101 INVITE
Timestamp: 1548963287
Server: Provider/2.0
Content-Length: 0

```

- A. voice class sip-profiles 2
request INVITE sip-header To modify "sip:(.*)@" sip:8005532447@
- B. voice translation-rule 3
rule 1./ /8005532447/
- C. voice translation-rule 4
rule 1/

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